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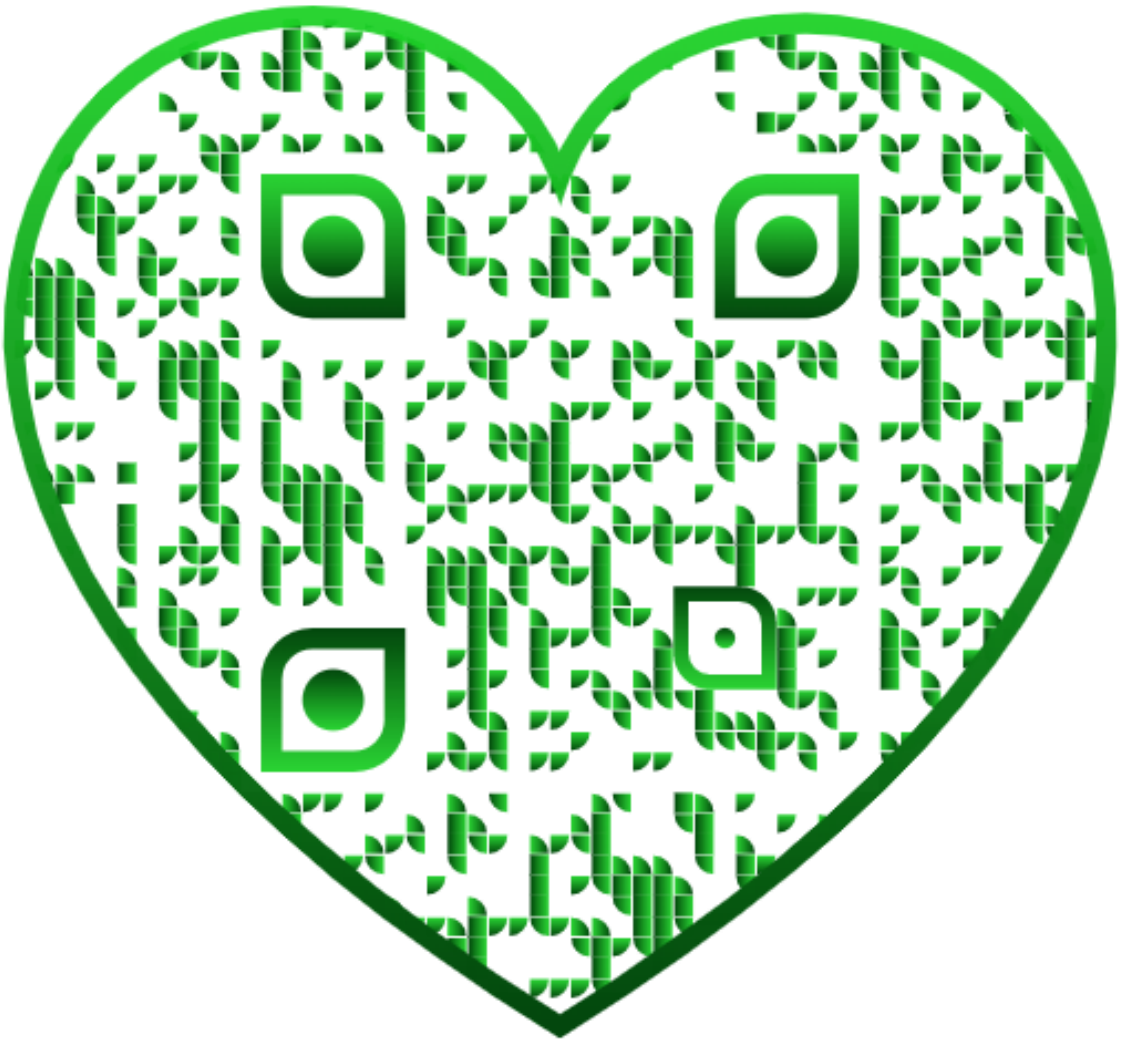

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Having teenage children is a great way to stay in touch with the next generation. Whether you like it or not, you're going to be exposed to current tastes in music, TV, clothes, reading material, social media, food, comedy, you name it. And Christmas and birthday lists will make it pretty clear what cutting-edge technology is currently desirable. So imagine my surprise when my eldest's number one request for Christmas 2023 was... a CD player.

This isn't some sort of kitsch '80s retro fetish. Like most people his age, he's perfectly comfortable with modern technology, and mostly plays music from his phone over Bluetooth. But he really wanted a CD player — not to mention a selection of discs spanning about six decades of recorded music. (I wish I could claim credit for his eclectic tastes, but I'd never heard of most of them, either.)

With vinyl sales also at a 30-year high, I wonder if this means that the streaming revolution has reached a natural limit. Perhaps the idea of listening to an entire album all the way through, with no adverts, isn't just some dated boomer idyll, but an inherently satisfying way of appreciating music. Perhaps it's human nature to want to have the art that means the most to us embodied in physical artifacts, not merely as digits in the cloud. And perhaps this is a trend that applies not only to music consumption, but also to its production.

The fact that music can be made using software alone has led some people to predict that hardware will eventually wither away altogether. Older musicians, the argument runs, are attached to classic synths and studio gear because of its associations, not because of its intrinsic usefulness or creative potential. As the generations that grew up with the Beatles and Led Zeppelin die off, so too will the idea that you need expensive hardware to make music.

I'm not convinced. Artists from the '60s and '70s haven't become irrelevant to younger audiences; in fact, streaming has proved to be a fantastic discovery medium for older music. The convenience of streaming hasn't yet managed to kill off the physical album as an art form. And laptop production isn't making traditional recording redundant, any more than sampling has wiped out instrumental skills, or video has replaced live performance.

No doubt some producers will be glad to ditch all of their hardware and do everything on a laptop. Others will stand out by doing everything the old-school way. But the best producers will be those who recognise that different approaches have unique strengths, and who can integrate the best of all worlds.

Sam Inglis
Editor In Chief

"Perhaps it's human nature to want to have the art that means the most to us embodied in physical artifacts, not merely as digits in the cloud."

SOUND ON SOUND

SOUND ON SOUND LTD (HEAD OFFICE)
ALLIA BUSINESS CENTRE
KING'S HEDGES ROAD
CAMBRIDGE, CB4 2HY, UK
T +44 (0)1223 851658
sos@soundonsound.com
www.soundonsound.com

WWW.SOUNDONSOUND.COM/SUBSCRIBE

WORLDWIDE EDITIONS

NORTH AMERICA

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EDITORIAL
sos.feedback@soundonsound.com

Editorial Director Dave Lockwood
Executive Editor Paul White
Editor In Chief Sam Inglis
Technical Editor Hugh Robjohns
Reviews Editor David Glasper
Reviews Editor Matt Houghton
Reviews Editor Chris Korff
Production Editor Chris Korff
News Editor Luke Wood

SUBSCRIPTIONS
subscribe@soundonsound.com
www.soundonsound.com/subscribe

Circulation Manager Luci Harper
Administrator Nathalie Balzano

PRODUCTION
graphics@soundonsound.com

Production Manager Michael Groves
Designer Alan Edwards
Designer Andy Baldwin
Designer Mick Reilly

ADMINISTRATION
admin@soundonsound.com

Managing Director/Chairman Ian Gilby
Editorial Director Dave Lockwood
Sales Director Robert Cottee
Marketing Director Paul Gilby
Finance Manager Keith Werthmann

ONLINE
support@soundonsound.com

Digital Media Director Paul Gilby
Design Andy Baldwin
Web Content Editor Callum Hall
Web Editor Adam Bull
Podcast Production Manager Atheen Spencer
www.soundonsound.com
twitter.com/soundonsoundmag
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DISTRIBUTION
distribution@soundonsound.com

International Distribution
Magazine Heaven Direct
www.magazineheavendirect.com

ADVERTISING
usadsales@soundonsound.com

Sales Director Robert Cottee
North America Sales Manager Dan Brown
Regional Sales Manager David Carson

MARKETING
marketing@soundonsound.com

Business Development Manager
Nick Humbert

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Steinberg Cubasis 3.6

DAW Software For iOS & Android

Steinberg's mobile DAW just keeps getting better.

JOHN WALDEN

Steinberg have shown an admirable commitment to mobile music production technology since they launched Cubasis for iPad, and the arrival of v.3.6 brings plenty of new features for users on both iOS and Android. Since *SOS* reviewed v3.2 in the May 2021 issue, a number of significant additions have appeared, including support for Chrome OS, Ableton Link and improved access to devices such as AirPods and other Bluetooth hardware. Alongside a whole host of efficiency and stability tweaks, plus the addition of a 'dark' keyboard display mode and a system for 'favouriting' preset sounds, the obvious highlights within this 3.6 release are four new instrument sound sets. Let's take a look.

LoFi Piano

One of these will be instantly familiar to users of Cubase on the desktop: LoFi Piano. As the title suggests, LoFi Piano features a series of piano-based sounds. These include some very respectable conventional pianos, but also plenty of processed variants with various degrees of lo-fi sonics. The UI is both simple and effective, with six key controls — Flutter, Compress, Saturate, Reduce, Filter and Reverb — that allow you to tweak the sound to taste. Having been a fan of this library on the desktop since it was first released, to my ears at least, the sounds here are pretty much identical, which is to say they are great: full of character and very usable. Oh, and it's a free download from within Cubasis 3.6, so what's not to like?

New IAPs On The Block

The other three instrument collection are all available as optional in-app-purchases. They are the HALion Sonic Collection IAP ($19.99), FM Classics IAP ($13.99) and Neo FM IAP ($13.99). The first of these provides a massive (over 1100 individual instruments) collection of sounds drawn from the desktop HALion Sonic instrument. These span a huge range of categories covering orchestral sounds, guitars, basses, drums, percussion, sound effects, pianos, organs, a range of synth-based instruments, voices and ethnic/world instruments. There are some fantastic sounds here that make something of a mockery of the compact form-factor of your average tablet. For example, try loading a patch such as Backing Section (a lush sustained strings patch from the Strings category) and using the touchscreen Chord Pads to trigger a few chords; it's as beautiful as it is epic. Whatever style of music you create, there is something for everyone here. Again, the UI works brilliantly for sound editing from a touchscreen.

The free Lo-Fi Piano expansion is included within the Cubasis 3.6 update.

I suspect the two FM IAPs are primarily derived from the desktop FM Lab expansion available for HALion. The FM Classics provides you with all the sounds from the classic DX7 and TX81Z synths. These are the sounds of the 1980s but also today, given how popular a synthwave flavour is in modern pop music. For a more modern take on FM synthesis, the Neo FM IAP is a great choice. If your home ground is modern pop or electronica, this will give you a top-notch palette of sounds to get creative with. Oh, and incidentally, Cubasis projects imported into a suitably equipped desktop version of Cubase will automatically get an suitable instrument match. Very neat.

Conclusion

The arrival of Apple's Logic Pro for iPad has undoubtedly provided some healthy competition for Cubasis and, if you use one of these on the desktop, it makes sense to stay 'on brand'. Seeing both Steinberg and Apple take the mobile platform quite so seriously is undoubtedly a measure of just how capable it is. Feel free to reminisce about the good-old days of cassette four-track recorders as a first step into the wonders of multitrack recording if you wish, but if you think of that old four-track as the original (undoubtedly revolutionary for its time) Ford Model T, then Cubasis 3.6 on a modern iPad is more akin to the Starship Enterprise. Yup, 2024 is a heck of a point in time to be starting your recording journey. Cubasis 3.6 is a mature, slick and feature-rich virtual studio that you can carry with you anywhere, and the new IAPs all provide impressive sound expansions available directly within the app. This is genuinely powerful stuff. ■■■

summary

Cubasis is an impressive illustration of just how powerful music production can be on a mobile platform. The IAPs added in v.3.6 expand the possibilities even further, with some excellent new virtual instrument options.

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—**Brandon Hickey, MIX Magazine**

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sE Electronics BL8

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NEIL ROGERS

Boundary microphones, or pressure zone mics (PZMs), are often overlooked in the studio. Typically associated with live sound, where their high SPL tolerance can be valuable, boundary mics can be placed inside, or fixed to the side of instruments, and so reach places that other mics can't. Although they're not things I use on every session, I've found some great applications for boundary mics in my studio — on piano in particular — and I was keen to see what this new release from well-known mic company sE Electronics could bring to my recordings.

Overview

A boundary microphone is designed to be positioned flush, or very close, to a flat surface — usually a floor, wall or ceiling. The close proximity of the capsule to a hard surface delivers two important benefits: reduced comb filtering (because the microphone only picks up what hits the surface, rather than any reflections coming off it), and increased sensitivity (because boundaries are where sound waves reach their maximum pressure).

The BL8 is phantom-powered and uses the same small-diaphragm capacitor capsules as the company's sE8 and sE8 Omni models, which means it can accept either cardioid or omnidirectional capsules. I was given both options with the review mic; swapping them out was reasonably quick and painless using the included miniature screwdriver.

The BL8 is a weighty, reassuringly solid and sleek-looking microphone, and it ships with a nice red leather case to keep it dust- and scratch-free when not in use. As boundary mics are typically placed or mounted on a flat surface, the BL8 caters for either scenario with both a non-slip rubber base and well-thought-out holes for attaching the mic to a wall or ceiling within a room, or for screwing to the lid of a piano, the side of a cajon, and so on. Another very popular placement for boundary mics is inside bass drums, and sE are keen to point out that the BL8 is compatible with the popular Kelly Shu Flatz microphone mount system, which »

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The base of the BL8 hosts the mic's pad, filter and EQ switches.

» allows a boundary mic to be suspended and vibration-isolated within a kick drum. In another nod to this popular application, sE have included Classic and Modern EQ options. Activated by a small switch on the base of the mic, these are aimed at sculpting the sound of kick drums, respectively offering a mildly or more aggressively 'scooped' tonality compared with the default sound. Lastly, there are switches for engaging a high-pass filter at either 80Hz or 160Hz, and the option to pad the signal by 10 or 20 dB.

In Use

My very first experiments with home recording involved a cheap boundary mic mounted on the ceiling above my drum kit in our terraced shared house (I feel bad for my neighbours in hindsight!). So it seemed fitting to start my review with drums, and before I looked at putting the BL8 inside the kick drum I experimented with a few different positions around my drum kit to see how the mic would fare in my studio's live room. I found myself liking it placed on the floor about three feet back from the kick drum, where it worked great as a kind of hybrid kick/room mic that proved very usable in a mix.

Many potential users will be looking at a mic like this with bass drums in mind, and I'm happy to report that the BL8 does not disappoint. I left the mic in its 'neutral' EQ position and positioned it inside the shell about six inches away from the beater head of the drum, on top of the soft cushion that lives inside my 22-inch Rogers kick. This gave me a plain but very focused capture of the kick drum that responded very well indeed to EQ. I then experimented with the mic's onboard EQ, discovering that the Classic mode adds a nice boost at around 60Hz, combined with a cut around 300-400 Hz. To my ears, the Modern setting is just a more pronounced version of the same curve, but both settings worked very well and were nicely tuned in to my idea of a good kick drum sound!

Boundary mics can be very effective for recording pianos. My piano technician friend once showed me his technique for using a pair of the cheap Realistic PZM mics commonly found on eBay — taped to the inside of the removable lower panel on an upright piano — and I've had a pair permanently in situ since. I was keen then to hear how a more 'hi-fi' boundary mic option would sound in this setting, and whilst it would have been nice to try a pair to cover the full range of the piano, the BL8 did a great job as a mono option. Tonally, I didn't perceive a huge difference compared to my cheaper in-house options, but I liked the fact that the mic always felt comfortable and not prone to any distortion if the player started hammering the keys.

Another use I found for the BL8 was recording an acoustic guitar, with the mic placed on a small table next to the player. It did a nice job of capturing a bright, clear guitar sound that provided a different perspective to my typical mic setup. The BL8 also sounded superb placed on the floor just in front of a loud bass cabinet, where it produced everything you would want for a loud rumbly bass sound with a clear and full low end. More than once I preferred the boundary mic option over my usual Neumann U47 FET positioned close to the speaker.

Summing Up

The BL8 is a rugged, stylish boundary mic that seems very competitively priced compared to the other high-quality options available. Capable of handling very high sound pressure levels with ease, the BL8 proved a very handy addition to my studio's mic options during the review period. The ability to just quickly put a mic like this on the floor in front of a bass cab or drum kit — and get good results — was great, and I often found myself preferring the BL8 over mics costing 10 times the price.

I was also reminded of just how good this type of mic is for getting usable room and ambient sounds out of less-than-stellar spaces. If you're recording drums in a small room, for example, you can open up some very creative options by placing a boundary mic on the floor, wall or ceiling.

Summing up, then, If you're on the lookout for a dedicated boundary mic, this seems a very good choice. Whether you're after a solid performer inside a bass drum, or you're interested in experimenting with what this type of mic can offer elsewhere in the studio, the BL8 comes highly recommended. ■■■

Omni Or Cardioid?

I made a point of trying both of the available capsules for the review, and not surprisingly, the main difference was that the cardioid capsule was more directional! Either capsule would work great for most of the applications I tried it on, although you would probably be better off with the more focused cardioid pattern if eyeing it up as a dedicated kick-drum mic. If you're looking for a more flexible option, or want to be able to capture conversations with multiple voices, or ensemble performances, the omnidirectional capsule would be the better bet.

summary

A classy, well-made and versatile mic that sounds great on a wide range of sources. Well worth considering as an alternative to the usual suspects!

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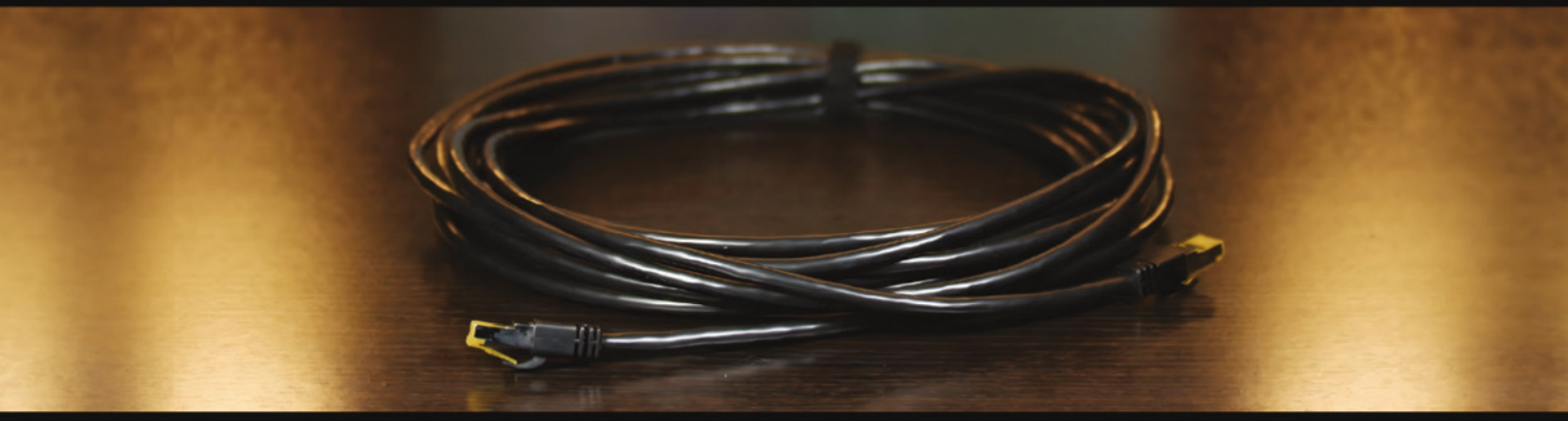

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Accentize dxRevive Pro

Dialogue Restoration Plug-in

MATT HOUGHTON

I loved Accentize's DeRoom Pro 2 — a slick, intuitive, machine-learning based reverb removal tool for dialogue that's capable of great results (https://sosm.ag/accentize-deroom-pro) — and their dxRevive Pro is every bit as impressive. It has a more a more ambitious aim, though: to extract "studio-like recordings from any source material". We're still talking specifically about dialogue, but as well as suppressing 'room tone' it tries to remove noises and artefacts from compression codecs, while also improving the spectral balance, by applying EQ in real time and synthesising new content to counter the effects of filtering, such as the band-limiting used for phone calls.

Overview

As with DeRoom Pro, dxRevive Pro is authorised by iLok and available in the usual formats for Mac and Windows. There are two different algorithms to choose from, one main 'amount' control (to apply as much or as little of the processing as you wish), and input and output level controls, each with a meter. An on/bypass switch is joined at the top by a preset menu and A/B compare buttons. In the middle, the source and processing are visualised, and it's possible create up to four frequency bands for which the amount of processing can be adjusted individually.

First, you'll need decide which algorithm to use. The default is Studio, in which the plug-in does it all: de-reverb, noise suppression and source enhancement. The other, Retain, focuses on tackling noise and artefacts, and drops the de-reverb side of things. Unlike something like Descript's Studio Sound process, this is a regular insert plug-in (it operates in real time, with a little latency) so you can run several instances in your DAW or NLE project, and can refine the settings after applying further processing — important, since compression and limiting often make artefacts more noticeable.

On Test

Inserting dxRevive Pro for the first time and turning the amount knob brought a smile to my face! A reasonable amount of care had been taken to set up mics for the recording, but the extraction of a nice, radio-style dry vocal was instant and nothing short of superb. But later, it allowed me to meld some very disparate voices, captured remotely over some pretty crappy mics in some very different spaces, into a coherent, intelligible discussion show: by comparison with the source files, the voices all sounded full, dry and intelligible. Where the speed of result is as important as the quality, then,

Can cleaning and enhancing dialogue really be this easy?

it could prove a great option. There's perhaps a little too much latency to be able to use it live when feeding a PA or monitor mix, but it could come in very handy for, say, journalists and news broadcasters for example, or on a streaming feed. Used 'off the bat' in this way, dxRevive Pro will also deliver results that will please the average self-producing podcaster wanting to clean up guests' phone contributions, or those looking to make the best of the audio from on-camera or phone mics used when shooting videos for YouTube, Tik Tok and the like. If that sounds like you, you may wish to check out the more affordable dxRevive (without the 'Pro'), which offers only the Studio algorithm and lacks the multiband facility and a few more minor features.

While dxRevive Pro is shockingly good and can make pretty much any dialogue recording cleaner and clearer, it can't turn everything into gold in an instant. You'll sometimes need to work harder on the multiband settings, or use dxRevive alongside other processing to get the best results. In particular, on poor recordings from VOIP systems, I could often hear artefacts in the high frequencies, where dxRevive seemed to be reacting to the effects of filtering. Reducing the amount of processing in just the high band often helped address such gremlins. Also, a little spectral de-noising at the outset in iZotope RX10 usually enabled me to coax better results from dxRevive generally. Importantly, though, these were better results than I could attain using RX10 alone. It's also worth noting that dxRevive can't undo the sort of overcompression borne of some VOIP services' auto-gain settings.

Retain mode should not be overlooked, particularly if you own DeRoom Pro, since the latter is a touch better at the specific job of reverb removal (at the cost of greater latency). To be clear, dxRevive is very good at this, but DeRoom Pro is better and having separate control over the noises and reverb can often be helpful.

While I might sometimes choose to combine dxRevive with other tools in search of the very best results, it really is a revelation: no other single processor I've used to date has enabled me to 'clean up' dialogue recordings anything like as well as this one. ■■■

summary

A stunningly effective noise-plus-reverb removal and enhancement tool for dialogue.

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Channel Strip Plug-in

Harrison's console-derived channel strip plug-in majors on speed and immediacy.

The channel strip as we know it was arguably invented by Solid State Logic, who added dynamics processing to the EQ that was already ubiquitous on mixing console input channels. The digital revolution took things to a new level, and even modestly priced live-sound mixers these days have per-channel compression, expansion/gating, de-essing and so on in addition to comprehensive EQ. Many software recording packages likewise have extensive processing built into every channel in their virtual mixer, and if they don't, it's probably included as an optional plug-in. So is it worth adding a third-party channel strip plug-in to your DAW's built-in resources? And if so, should that plug-in be Harrison's new MPC Channel Strip?

It's certainly a plug-in with pedigree. Harrison's MPC consoles are super high-end digital powerhouses targeted at the world of film dubbing, expandable to gazillions of channels and with enough DSP to compute the answer to life, the universe and everything. Every aspect of the feature set, including the channel processing, has been finely tuned through years of feedback from professional users. They are tools designed to do their job as quickly and efficiently as possible, whilst remaining sonically neutral — and now their core processing has been spun out as a plug-in.

Strip Mining

Authorised using the iLok system and available for all major native formats, the Harrison MPC Channel Strip plug-in has five basic processing elements. The bulk of the screen real estate is taken up by the equaliser, each of whose eight bands can have shelving, filter, bell, notch or 'search' responses. There's also a separate, dedicated filter section, which features two additional bands that can have either shelving, filter or notch shapes. In the lower half of the window, you'll find three separate dynamics processors: a compressor, a de-esser, and a "de-noiser", which is actually a frequency-conscious expander.

Eight EQ bands will usually be more than enough for both filtering and shelf/parametric tasks, but the additional pair of filter/shelf bands is nevertheless useful, not only because they have a wider choice of filter slopes but because the order of MPC Channel Strip's processing elements can be freely varied. You could, for instance, have the main EQ set post-compressor, but use the filter section before the dynamics. However, I was a bit surprised to find that there's no option to switch the filter section into the compressor side-chain, or use an external key input to trigger the dynamics.

Other than that, the compressor is certainly well specified. The ratio is continuously variable from 1.1:1 up to 100:1, and the knee and time-constant controls are equally flexible, with a programme-dependent auto setting available for the latter. A Depth control serves to limit the maximum gain reduction that can be applied: a very useful feature that should be implemented on more compressors! Threshold and make-up gain are set using sliders rather than dials, and between them is a gain-reduction meter. Unfortunately this has a fixed scale of 60dB, which means there's not a whole lot of meter action going on in normal use. (Side-chain filtering, an external key input and scalable gain-reduction metering are already available in Harrison's MPC Compressor plug-in, and the developers are working to add them here too.)

Sibil Engineering

Harrison's de-esser is one of the highlights of this plug-in, being distinctive, highly effective and a breeze to set up. It's essentially a highly optimised two-band dynamic EQ, the idea being that you can target the middle band towards the most problematic sibilant frequencies and use the high band to apply more gentle treatment at the very top end. But you don't have to use it like that, and in fact it has applications that go well beyond de-essing. The mid band can operate right down to 200Hz, and the high band down to 2kHz, so there's plenty of scope for midrange tone-shaping on sources other than »

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» vocals. Again, there's a useful choice of real-time analyser modes, and you also have control over the attack time, but what stands out most of all is its ease of use. The control set just does exactly what you need, in a supremely intuitive way, with no superfluous parameters or information.

Like several other aspects of the plug-in, the "de-noiser" is said to be optimised for vocal use. The principle is easily understood: as the signal level falls below the threshold, attenuation is applied at either end of the frequency spectrum, hopefully minimising any background hum or hiss in a relatively natural way. There's no user adjustment of the time constants, the ratio, or the turnover frequency or shape of these bands: just sliders that set the threshold and the degree of low and high attenuation. But once again, what looks almost too simple on paper is highly effective in practice. Given enough time, I think most of us would turn first to advanced offline noise-reduction tools such as CEDAR and RX, but this produces instant results, and provided you're conservative with the settings, does minimal damage to the wanted audio. Once again, though, I found the metering problematic; especially on non-vocal sources, you can sometimes hear that the de-noiser is acting even when the meter suggests it isn't.

While I'm complaining about metering, it's also worth pointing out that there are no input and output level meters, nor any indication of overloads within the plug-in. Get too happy with the output Trim or the compressor make-up gain and you can certainly hear distortion, but nothing appears on screen to tell you that anything is amiss.

Smooth Moves

If this description has left you thinking "That's all very well, but look at the price! How can Harrison justify charging more than my DAW cost for a channel-strip plug-in?" it's a fair question, although it should be noted that it will be available at a much lower cost during sale periods. The appeal to those who already work on MPC consoles and want to be able to use the same tools and techniques natively is obvious. For the rest of us, the answer is mainly to do with ease of use. The sound is very clean and the feature set decent enough, but it's the user interface that stands out for its wealth of neat ideas and time-saving details. For example, you can use the mouse to pick up nodes in the graphical EQ display and move them around, with the scroll wheel adjusting bandwidth, as you'd expect.

Auto Solo is a clever and effective tool for helping to pinpoint the frequencies to which the de-esser should be targeted.

But each node also has a numeric display below, and moving the mouse pointer over one of the numeric fields and twirling the scroll wheel provides a very efficient way of fine-tuning your settings. The pop-up that sets the shape of each band can also be short-circuited by scrolling, which is handy, because it allows you to drop in and out of 'search' mode pretty much instantly. Setting a band to search mode essentially turns it into a band-pass filter, which you can sweep up and down in order to pinpoint that troublesome snare ring or vocal resonance. Once you've located the problem area, another brief twirl of the scroll wheel will set that band to notch or bell mode in order to dispense the right treatment. Equally thoughtful is the real-time analyser display that can optionally be set to appear behind the EQ curve. This offers four different modes, including the very intuitive 'lightning' option, essentially a spectrogram where peaks are outlined in a brighter white.

> **"The sound is very clean and the feature set decent enough, but it's the user interface that stands out."**

Similar ergonomic design features are apparent throughout the plug-in. The ear icon next to each processing section temporarily disables all the other sections, so you can hear what it's doing in isolation, whilst the eye-like symbol that appears at the top right of the plug-in interface is a momentary bypass, disabling the plug-in only while you hold down the mouse button. The de-esser has a clever Auto Solo function, which as the name suggests, solos whichever de-esser band you're dragging around at a given moment, then switches itself off when you let go. This makes it supremely easy to find and target the problematic sibilant frequencies in a vocal. And although it isn't possible to save presets separately for each individual section, circular arrows allow the settings to be switched back to the default at a single click.

So, although many of us already have very capable channel strip plug-ins bundled with our DAWs, it would be a mistake to dismiss MPC Channel Strip for that reason alone. I certainly found it more intuitive and faster to use than the free Avid Channel Strip plug-in you get with Pro Tools, for example, and I suspect the same would go for the equivalents in other DAWs. It's a big investment, but they say time is money — and, especially if you do a lot of work with vocal recordings, the ergonomic design of this plug-in could well save you enough time to recoup that investment.

summary

Although it's costly and would benefit from better level metering and side-chain access, Harrison's channel strip plug-in offers many ergonomic benefits, with its slick graphical interface making it a pleasure to use.

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MARK GORDON

Soniccouture AC-DR

Software Instrument

Soniccouture have combined acoustic and electronic percussion to create something truly original.

"A virtual Instrument designed to emulate classic drum machines of the past" is definitely a remit that has been explored far too many times for anyone to be remotely interested. Unless, of course, there is a significant new twist on the idea — and I might just have one for you. Soniccouture have taken samples of acoustic drums and percussion instruments, recorded in new and innovative ways, and designed a hybrid instrument that recreates the sounds and styles of the drum machines of the past. Acoustic drums that sound electronic — now that *is* an interesting new twist...

Turn The AC On

AC-DR is a Kontakt Player instrument. This means that you do not need to own the full version of NI Kontakt to use it, and it will run as a plug-in instrument inside any compatible host program.

The instrument itself is split into two main screens: a Drum Machine overview of the individual elements, along with their edit parameters; and a Beat Tools page that features three unique rhythm generators. It comes with a number of preset kits and rhythms that help you get an idea of what this virtual instrument is capable of, but the real fun is in creating your own sounds and experimenting with the beat-creation tools.

Focusing on the Drum Machine screen, a familiar layout features 11 large, coloured blocks that represent Bass Drum, Snare Drum, Rim Shot, Tom Toms, Hi-hat, Ride and Crash Cymbals, Cowbell, Tambourine and Shaker. Each block includes its own level fader, pan control, and mute and solo facility, plus several knobs that represent the differently recorded 'channels' of that instrument. For example, for the Bass Drum the knobs are Beater, Sub, Knock, Trans and Space. By turning up the Beater knob, you introduce a sample recorded with a mic pointing directly at the batter head of the kick drum, while the Sub sample was recorded using a Neumann FET47 mic in front of the drum. The Knock sample was captured using an UKKO contact mic attached to the shell of the drum, which produces a thin and clicky quality, and the Trans element is the result of the whole sample being passed through an Overstayer Saturator, introducing a level of dirt and distortion. Finally, there's the Space knob, which is particularly interesting, as its samples were created using the reverb chambers at Rockfield Studios.

Each sample has its own character, and when used in combination with each other, they can create the most amazing and varied drum sounds. Of course, this method isn't limited to the kick drum. All of the drum instruments have unique combinations of samples, recorded using a wide variation of beaters, microphones and analogue processing. The way the different channels can be blended and manipulated reminds me of the Nord Drum, which uses click, tone and noise elements that are ultimately combined to produce the final drum sound.

Clicking on the drum name fills the space in the centre of the screen with a very extensive editing window. An almost infinite level of sound manipulation is offered. Various drums, cymbals and percussion instruments were used in the recording process, and here you can pick any of them — from 18-inch to 26-inch bass drums, wood and metal snare drums, 12-, 14- and 16-inch hi-hats, crash »

Soniccouture AC-DR
$149

PROS
- Innovatively recorded samples.
- Almost limitless editing options.
- Excellent beat creation tools.

CONS
- None.

SUMMARY
A hugely innovative, unique twist on the virtual drum machine that is creative and addictive in equal measure.

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UNIVERSAL AUDIO

Beat Shifter looks like a standard grid editor but adds the ability to shift the location of individual beats in random but musical fashion.

» and splash cymbals recorded with contact mics, and tom-toms struck with sticks and mallets. It's worth mentioning at this point that there are 10 round robins per hit, so as you play repeated drums, a different and unique sample plays with each hit, creating an organic quality reminiscent of drum machines like the Roland TR-808, where each beat is uniquely created by the analogue circuitry. You can also turn off the round-robin feature to achieve the effect of the same sample retriggering, giving the 'machine-gun' effect of a Linn Drum or DMX.

Parameters available in the edit window include Filter, Envelope, Compressor, Transient, EQ, FX and Pitch, and these can be manipulated per sample channel. What do I mean by this? Well, you can apply any of the edit parameters to any of the samples. For example, the Snare includes Body, Rim, Wire and Space samples. By clicking on the numbered buttons that relate to each sample, you can give the snare Wire more decay while reducing the attack on the Body sample, changing EQ on the Rim and extending the decay on the Space sample. It really is quite mind-blowing. The pitch, filters and envelope settings also enable you to create some very interesting bass sounds using the tuned toms.

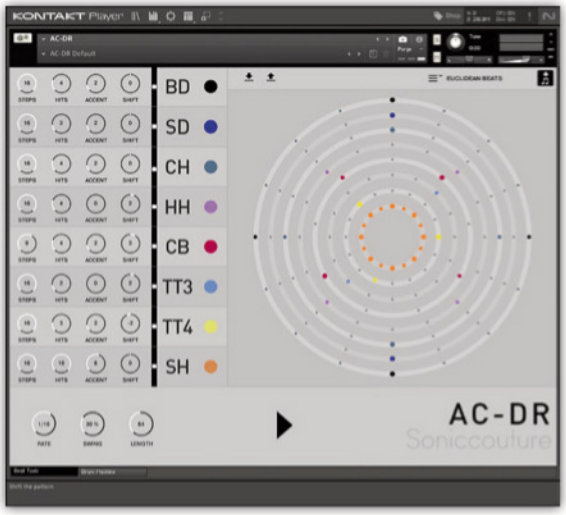
In addition to the individual edit parameters there are two global effects processors, featuring a convolution reverb and digital delay, plus a Master channel that offers three Bus compressors, Limiter, EQ and Tape Saturation.

Fascinating Rhythm

The Beat Tools screen will be familiar to users of Soniccouture products, as it appears on most, if not all, their percussion-based virtual instruments. Featuring three different editing options — Beat Shifter, Euclidean Beats and Poly Beats — this screen provides some very innovative methods of creating rhythms.

Beat Shifter looks like a conventional pattern grid, with eight lanes that can be assigned to any of the 16 available instruments. Clicking on one of the steps adds a beat, as you would expect. However, for each drum lane you also get Shift, Step, Direction, Velocity and Chance Of sliders, which, in very simple terms, move the beats around the grid in a random fashion. It sounds odd, but it can create amazing rhythms that you'd never come up with by programming manually. The beats continue to change and evolve through each cycle, but if you hit on something you like, you can Freeze the beat and export it as a MIDI file. Press Freeze again and the beat continues to evolve.

Poly Beat offers a similar grid-based layout, but in this case each of the eight lanes can have a different number of beats, ranging from one to 32, plus everything in between. Again, you can create some amazing polyrhythms by combining different numbers of beats within the same pattern. A nice way to work is to 'lock' two or three elements, such as kick and snare, to a grid and then use the Randomise button to manipulate the other drums over the top of that solid beat.



Eucildean Beats offerers a unique way to create grooves and rhythms 'on the fly' by manipulating the number steps, hits and accents of each drum using a set of virtual knobs.

Finally, Euclidean Beats is a drum editor format I was unfamiliar with. The eight drum lanes are laid out in a circular pattern, and virtual knobs to the left let you specify the number of steps, hits and accents for each drum. If you assign MIDI controllers to the various knobs and use an external control surface, this editor does lend itself well to the real-time manipulation of beats. Like many musicians, I often end up playing beats into my DAW and creating parts manually, but the Beat Tools provided by AC-DR take this to a different level that simply isn't possible using a normal drum editor. All in all, this is an exceptional and unusual set of editing tools that'll keep you creating new beats for years to come.

Can't Beat It

I absolutely love AC-DR! All the sounds have been created organically, whether recorded with microphones, passed through an analogue processor or given character in a physical space. The raw sound sources are beautifully recorded, and the various methods and ingenuity used to create them results in a huge palette of unique material to work with. The way in which each element can be manipulated beyond imagination is endlessly creative and incredibly addictive, and this is where the lines blur between acoustic and electronic. The fact that the instrument is based entirely on samples of natural acoustic drums appeals to the drummer in me, but you can also create the most remarkable 'electronic' sounds. In combination with the comprehensive beat creation tools that lend themselves to the more mechanical side of drumming, the AC-DR can indeed "emulate classic drum machines of the past" but it is much, much more than that — and very much worth adding to your VI arsenal.

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Moog Mariana

Software Synthesizer

Moog's new bass soft synth goes deep. Really deep...

WILLIAM STOKES

Available for macOS and Windows in VST3, AAX and Audio Units formats, the new Mariana soft synth has been described as "the Moog that never was". It is dubbed a 'bass synthesizer', hence its being named after the deepest place on Earth.

Back To Bass-ics

Mariana has more than a few tricks up its sleeve, but it is at its core a fairly simple subtractive monophonic synthesizer. The SYNTH 1 page presents two oscillators offering sine, triangle, triangle-saw (or 'sharktooth'), sawtooth and square waves — with the chosen shape only applicable to both oscillators together. The oscillators' mutual waveshape can have its shape edited by a Duty Cycle knob, which is essentially a pulse-width control, but for all the wave shapes, not just the square wave.

Familiar territory so far, but there are a few particularly well-suited controls for bass here as well. One such is a Key Reset button, which helps maintain a consistent phase relationship between the two oscillators by having them reset with each new key, hopefully ensuring that different notes and pitches sound as similar as possible. There's also a knob to adjust oscillator 2's phase relationship with oscillator 1. There's a sub-oscillator with variable octaves and three simple waveforms to choose from (sine, sawtooth and square), as well as a control to adjust its own phase relationship with the primary oscillators. There's also a variable noise generator, with an accompanying knob to cycle through red, pink, white, blue and purple noise variations. The sub has its own multimode filter, while the other oscillators have the useful pairing of both a high-pass and low-pass filter, which can be routed in series or parallel. The noise can also be high-passed by itself to float above the rest of the synth voice: a nice touch for adding a percussive edge to otherwise murky low-frequency information.

Next along is the CNTRL 1 page, for all things movement and modulation. It contains three fairly simple LFOs and three envelopes, including an assignable MOD envelope that has additional Delay and Hold stages compared with the other two's ADSR stages. Finally, there are two random generators for rate-variable or sync'able stepped or smooth random values; the latter uses a Perlin noise generator, so named for its inventor Ken Perlin, and is comparatively natural in feel. The random sections also have optional slew, which is a handy addition. The modulation routing itself takes the form of a drop-down menu: click a knob, click the modulation source and adjust the movement range with useful, colour-coded visual cues.

Doubling Up

Here's a question: what's better than a synth with all the features above? That's right: two synths with all of the features above! Yes, Mariana has two identical layers of architecture — SYNTH 1 and CNTRL 1, and SYNTH 2 and CNTRL 2 — and this really does expand its functionality by more than just a factor of two. One one level, it being a soft synth, you might argue that you could just instantiate multiple instances of Mariana, but beyond the CPU question (Mariana does take its toll), there's another benefit to this: you can modulate parameters on either layer from either CNTRL panel, mix and match sub-oscillator responses, blend waveforms together, and legions more. It's also possible to arrange the two layers into a duophonic synth.

"Hold on," I hear you say. "It can only go up to duophonic?" That's right. "But it's a digital synth. What's to stop it from being fully polyphonic?" You may have been thinking similar thoughts about several of the functions I've mentioned. The fact that there are only three wave shapes for the sub oscillator, for example, or that both oscillators on each layer can only have their shapes adjusted together. "We're in software now, where the boundaries of hardware that infuriated musicians in years gone by need not apply! Why all the limitations?"

Endless features in software might sound good on paper, but they do tend to create option paralysis. In my humble opinion, the more developers can work against this, the better. I do not, for example, think it's helpful that Arturia's software emulation of the Minimoog Model D (to name but one) makes it fully polyphonic, because its monophony is »

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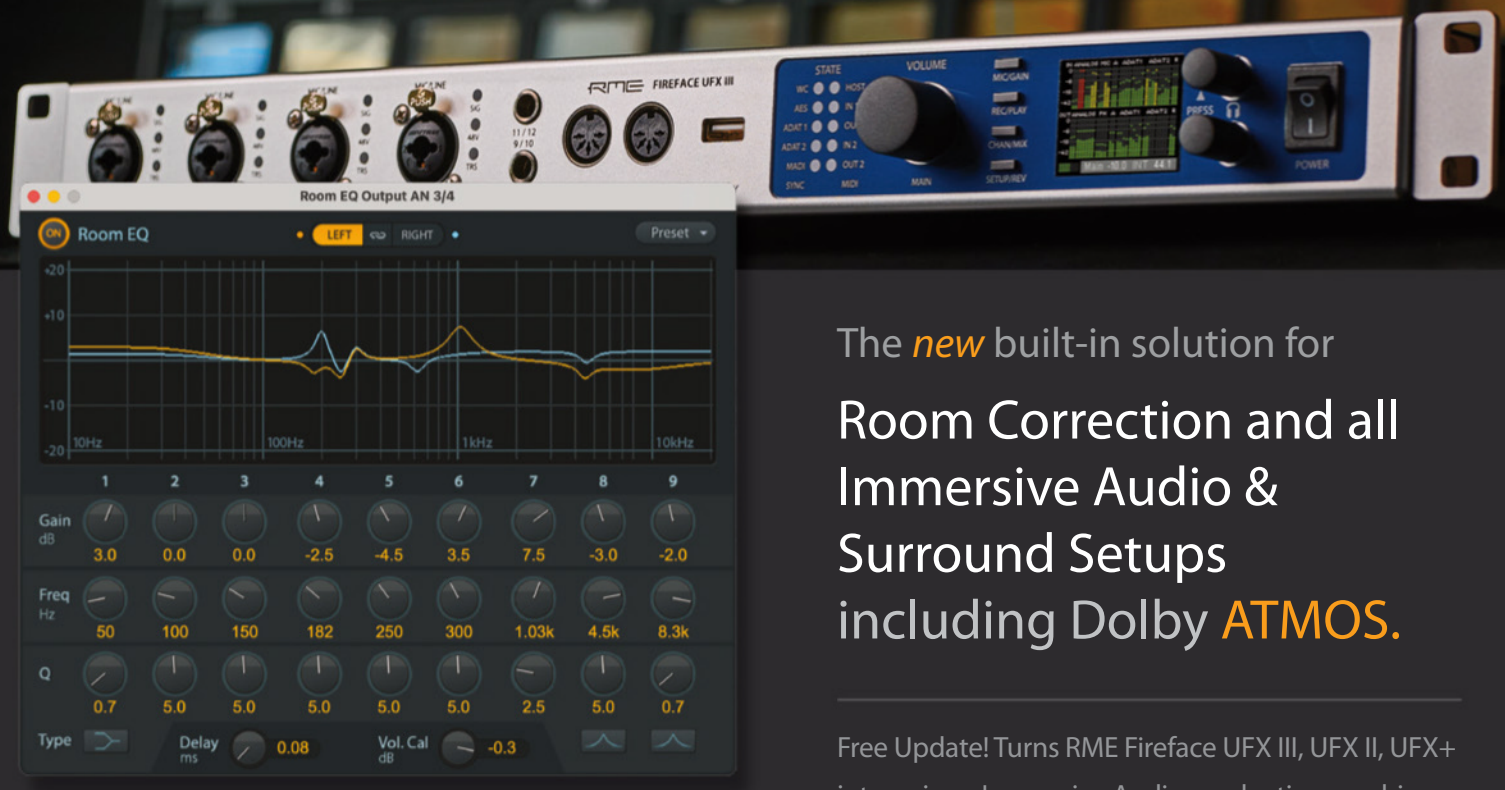

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precisely one of the things that means players have used it in a particular way, with more emphasis on sound design and the significance of singular musical gestures. In this respect, Mariana does a brilliant job, and it's one of the reasons I think the observation of it being the "Moog that never was" is a good one. Its workflow really does make it feel like an emulation of a hard synth — just one that doesn't exist. It's a software synth for those, like me, who approach software with a degree of trepidation. It incorporates actual design decisions to guide the user, which is much more interesting than simply throwing everything at the wall and making every single value adjustable down to the last, undetectably insignificant increment.

The OUTPUT page brings the two synths together, but with different effects.

Sum Of Its Parts

It's on the final page, OUTPUT, that Mariana's two synth layers deviate, but only in their effects. SYNTH 1 is given a delay line, while SYNTH 2 is endowed with decent enough chorus. One area in which Mariana performs very well is its imaging: here there's a lot of room to play with the stereo field, with effects tangibly widening their respective voices and both voices independently pan-able, on top of their per-voice oscillator Spread settings.

Beyond this, the OUTPUT page has a host of functions that perform surprisingly well. With software instruments, developers are invariably faced with the question of what to include natively and what to leave up to other plug-ins. Mariana has gone for goal with on-board delay, chorus, saturation for both layers and even a compressor. With the delay and chorus working across their two voices independently, and the option for the saturation to also work as such, these ultimately come to feel more like contributions to the synth voice itself than effects — particularly since they're modulatable from within Mariana's matrix and can therefore participate in the movement of other aspects of a patch. Thankfully, these effects genuinely sound good, so I wasn't left wanting much in this department. The compressor, meanwhile, won't wipe the floor with your Fabfilter Pro-C, but it's not trying to. In practice it's just a quick and useful stereo bus compressor, for melding what can amount to a lot of movement and disparate information into a more coherent singularity. This certainly makes Mariana easier to work into a mix, and it means that any other dynamics processing you choose to put on its channel strip can both focus on more creative gestures and keep your plug-in list that little bit more manageable.

If you are after additional effects, Moog have made it very easy to integrate Mariana with none other than their own range of Moogerfooger plug-ins, courtesy of a virtual CV system that plumbs right into Mariana's modulation matrix. While the workflow here isn't the most intuitive, it is capable of some great results — and the more outré the Moogerfooger the better. Rack up enough of them and you can create something of a virtual modular setup right there in your DAW.

One wonders if Moog are turning their attention to the development of a more fleshed-out digital ecosystem; it's no secret that changes have been made since the company joined the InMusic conglomerate, so I would hazard a guess that we can expect more software goods from the company in the near future. In fairness, if they're anything like this then there'll be very little to complain about. Mariana manages to retain as much Moogness as you could ever hope for from an original soft synth, and with such a nicely balanced set of options and limitations, not to mention a formidable sound, will be welcomed with open arms by any lover of Moog hardware.

Each synth has a CNTRL page featuring LFOs and envelopes.

summary

A solid, well designed synth bringing all the classic Moog bass your DAW could ask for.

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Native Instruments Kontrol S88

Controller Keyboard

NI's latest 88-note controller introduces a keyboard with hammer-action polyphonic aftertouch and a lot more besides.

ATHEEN SPENCER

There are a fair few weighted controllers available these days, but until now, there hasn't been one that features polyphonic aftertouch. Indeed, this is still far from universal even on synth-action controllers. So it was refreshing to hear that Native Instruments and Fatar have worked in collaboration to create a brand new keybed that provides polyphonic aftertouch for the weighted-action S88, which is otherwise known as the Kontrol MkIII keyboard.

First Impressions

Unboxing is sometimes challenging with 88-note keyboards, but the S88 was a lot easier than expected, as it weighs a very manageable 13.5kg. Not bad for a hammer-action controller, and something that makes it feasible to cart around to gigs.

At first glance it's very appealing. It feels sturdy, has a single-shell body with no seams, and a logical layout with buttons grouped by function. This model sports a single large glass display screen, rather than the two smaller screens offered by previous versions. There are eight soft buttons located above the screen, eight soft knobs below, and to the right is the 4-D Encoder, a multi-function control that can operate as a joystick, button and continuous encoder.

It's a relief to see a chunky pair of pitch-bend and mod wheels to the left of the keyboard, which I find much more practical than joysticks placed above the keys. Of course this is down to personal preference, but for this keyboardist it's a welcome change. The wheels each have an LED light, which is handy for locating them easily during performances on a darkened stage or in a mood-lit studio. The knobs and wheels are aluminium, and feel more sturdy and superior to those on the previous model.

Above the wheels is the assignable touch strip, set to CC11 by default but with the option to switch to Pitch-bend, Control Change or Program Change. Moving this to be positioned above the wheels on the MkIII makes logical sense, as it can be more easily accessed during a performance and is less likely to be triggered accidentally.

For those of you who like to balance additional kit on the edges of your keyboards, you'll be glad to know that there's plenty of space for this, with flat areas of approximately 17 x 40cm on the left and 17 x 45cm on the right.

On the back of the keyboard are the power switch and two USB-C sockets, one to host and one for mains power if needed. There are also MIDI In and Out sockets, which can be used normally or assigned to work as a MIDI interface with your computer. Four assignable pedal sockets are available, of which the first two are set to sustain and expression by default. Having such a generous complement is handy if you need to use soft, sostenuto, damper or any other performance controls.

Changes can be made to hardware and controllers from the settings button, with multiple pages accessible using the soft keys above and arrow buttons to the left of the screen.

Faders and pads are surprisingly absent. This was apparently done to keep down costs, and because Native Instruments have other products that handle things like Maschine integration, but it's strange nonetheless: composers, producers and performers all benefit from these controls, and it's often easier to have them all within one unit due to limited desk and stand space. Using the touchstrip to add expression can partially compensate for this omission, but otherwise you're left with the pedals or the soft controls surrounding the screen for all other functions.

The Fatar Keybed

While all three models in the new range (the S49, S61 and S88) are identical in terms of function, the S49 and S61 have a synth-action keybed (TP90), while the S88 benefits from hammer action (TP100).

Since Italian company Fatar have been designing keybeds for keyboard manufacturers since the late '80s, they know a thing or two about creating very playable instruments. This particular keybed for the S88 has been in development as a collaboration with Native Instruments since 2017, and has had a complete redesign to include polyphonic aftertouch in a weighted keyboard for the first time. This very welcome addition uses an FSR (Force Sensitive Reistor) matrix to sense independent key pressure without affecting the feel of the instrument.

Although the keys feel like sightly harder work to play than those of a higher-end digital piano, the weight and expression of the keybed are extremely impressive and should please those

Native Instruments
Kontrol S88
$1299

PROS
- Polyphonic aftertouch in a piano-action controller.
- Speedier workflow with Kontakt integration.
- Light guide for instrument ranges and keyswitches.

CONS
- Some troubleshooting required during setup.
- Lack of faders and pads.

SUMMARY
The combined playability and functionality of this controller make it a very worthy contender for a place as the hub of your studio.

keyboard players who have had some classical training. The velocity curve can be adjusted in edit mode, or alternatively you can hit the fixed velocity button to instantly play everything at 127.

All three versions come bundled with a selection of software that includes Ableton Live Lite, iZotope Nectar Elements, Komplete 14 Select and Komplete Kontrol. Some of these included instruments come with presets that use polyphonic aftertouch, so you can explore this feature straight away.

Getting Set Up

Although the S88 MkIII is advertised as plug-and-play, it took me a little while to get things fully up and running. First you need to download the Native Access app and register your serial number. Once that's done, you'll need to download and install the Hardware Connection Service software. This is a good time to check that Komplete Kontrol and Kontakt are up to date. Plus, if prompted by the hardware, you may also need to download the KSMK3Update app to make sure that you have the latest available version of firmware for your S88.

Next you'll need to map your DAW — make sure your S88 is switched on and boot your software. For Logic, an option to auto-assign controllers pops up, and for the sake of speed, I accepted, with a view to adjusting later. For other DAWs you'll likely need to complete setup manually, something that may possibly change with future updates. If at this stage you find that your pre-installed library products are not available, you'll need to relocate them by opening Kontakt as a standalone app.

Integration

While the S88 is a very appealing keyboard in it's own right, it's really designed for integration with the Native Instruments ecosystem. As someone who has always preferred working on a computer screen, I experience a slight reluctance when it comes to using smaller screens on hardware, but it soon became apparent why working in this way could be hugely beneficial.

While previous versions of the Kontrol keyboards have worked with Komplete Kontrol software, you can now also load Kontakt and access your entire Kontakt library. When activated, the library artwork is displayed on the screen. The knobs below the screen, meanwhile, are touch-sensitive and can be used to filter sounds by Brand, Product Name, Bank, Sub-bank, Instrument Type, Subtype and Instrument Character. Use knob 8 or the encoder to scroll through the displayed list of sounds and hear a preview without loading, then press on the encoder button to select. Once sounds are loaded, the soft knobs can be used to either edit the sound or apply filters during a performance.

Sounds can be layered quickly and easily by using the soft button at the top to select the next available sound slot. You can store favourite sounds for fast recall later using the relevant soft button, and save an edited or layered sound by naming it on your computer screen so that it will then appear within the user presets on your controller. Komplete Kontrol users can also load loops, one-shots and effects.

The DAW transport controls are the handiest item for speeding up your workflow. Hit the DAW button to the right to view your mixer on the screen and select your track using the joystick function on the encoder. Arm with the Record button, start recording with Play. Pressing the Stop button a second time returns you to the start of the track. You can even quantise and record automation directly from the Kontrol. While the whole process takes a little getting used to, a small amount of perseverance will see you through the learning curve, and it could ultimately save a lot of time to work in this way.

The Purpose Of The Lights

Most musicians and studio personnel that I know can never have enough lights in their studios, and the S-Series doesn't disappoint here. However, they're not just there to keep us entertained. Switch on Play Assist, set the key of your track and the lights offer a guide for playing various scales; while that may seem like a bit of a cheat, it could actually be used as a valuable training tool. Chords and arpeggios can be played from a single note based on your selected key, which is handy if you're not a keyboardist and need some quick inspiration.

The light display also shows the playable range of loaded instruments, groups drum sounds by colour, and indicates where keyswitches are, saving a lot of time.

Conclusion

So who is the S88 for? While it appears to be primarily designed for studio use by producers and composers, performers will also benefit from the sturdy design, weighted action and fast recall of user sounds. Faders and pads will be missed by some, but it's undoubtedly the introduction of polyphonic aftertouch into a hammer-action keyboard that's the highlight of this release, and the feature that will set it apart from other weighted keyboard controllers.

$ Kontrol S49 $749, S61 $849, S88 $1299.
W www.native-instruments.com

Steven Slate Audio VSX

PHIL WARD

Virtual Monitoring System

Can headphones and modelling software really replace a professional mix room?

The Steven Slate Audio VSX system caused something of a stir when it was first launched. The idea that they would bring an end to the issue of monitoring translation by promising "Perfect Sounding Speakers. Now In Headphones" seemed for many, me included, to be somewhat over-ambitious. However, the initial brouhaha has died down a little now, and a new version of the VSX software has recently been released, so it seemed time for *SOS* to take a look. Along with introducing some extra room models, the new release also previews an HRTF feature in two of the room models that is slated to be included globally in a future release.

Bundle Up

VSX comprises a pair of headphones and a software suite including AAX, AU and VST plug-ins, along with a newly introduced 'Systemwide' application that enables VSX to do its magic outside of a DAW, processing any audio that the host computer is playing. But what exactly is the VSX magic? VSX is a monitoring modelling system that aims to render, through its headphones, the sound that would be experienced in a variety of different locations or from alternative hardware: Steven Slate's mix studio, a car interior, some alternative headphones and a smartphone, for example. In the case of modelled studios (and club or car spaces), the principle is that VSX creates a binaural analogue of the environment and its audio hardware, allowing one to hear how a mix will sound in those locations when played through the monitors or speakers present there. As an extension of the technology, VSX also offers non-binaural models of some other generic headphones. VSX comes in two editions, Essential and Platinum, the latter providing access to an expanded list of modelled environments and headphones — 20 in total (with more apparently on the way), compared with seven for the Essential edition. Modelled environments can also be purchased »

Steven Slate Audio VSX
From $299

PROS
- Convincing room and monitor modelling.
- Potentially genuinely useful.
- VSX headphones are fundamentally competitive.

CONS
- None.

SUMMARY
Steven Slate Audio VSX is a fascinating and ingenious audio product that successfully achieves exactly what it sets out to do. I went from sceptical to convinced.

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» individually, so if, for example, you can't possibly mix without knowing how your work will sound inside a Tesla, that model can be added to an Essential VSX edition.

Can Do

Before getting into how well the VSX binaural modelling works, I'll look at the VSX headphones themselves. They are somewhat generic in appearance and style but nonetheless of high manufacturing and finish quality, with a nicely padded headband and generously dimensioned oval ear pads. The connection cable attaches via a 3.5mm jack to the left earcup, and the headband and ear pads are covered with faux leather. They are unremittingly black other than the grey VSX logo. As well as lacking colour, they also lack any of the luxury aesthetic touches that seem in recent years to have become *de rigueur* for 'audio wearables', but to my mind that's a positive; headphones intended for professional use are tools for a job, not fashion items.

The VSX headphones are of closed-back, circumaural design and feature a high-tech beryllium-coated diaphragm and a low-frequency porting arrangement called APS (Acoustically Ported Subsonics) that is described as a "sophisticated internal tuning vent system and patent-pending bass coupling for optimal low-end performance". I'm not entirely sure what that actually means in real-world audio engineering terms, but it appears to describe a headphone driver loading technique that at low frequencies allows the headphones to operate as a kind of hybrid between closed and semi-open. In any case, and owing to the circumaural design, low-frequency performance will depend to some extent on the quality of the air seal around the ears. The more generous the ear pads, the more consistent that seal is likely to be, and the VSX ear pads are generous.

In use, the VSX headphones are comfortable, with about average weight and ear-pad pressure. Comfort in use is important, because if the VSX proposition really does result in "Perfect Sounding Speakers. Now In Headphones", chances are you're going to be wearing them for extended periods.

Plugging In

Used conventionally, without the VSX app running, the VSX headphones perform reasonably well, with a wide bandwidth, accurately rendered and

extended bass, and a relatively neutral tonal balance (although not quite as neutral as my mid-price circumaural headphone reference, the AKG K371). At the same time, they sound to me a little lacking in the upper midrange, and display what I'd describe as a slightly nasal character. That said, all headphones, and especially closed-back types, display an aural signature to some degree, and the VSX are generally typical of the breed.

Such use is probably the exception rather than the rule, because with the VSX Systemwide app running it's likely that most of the time the headphones will benefit from the EQ that the app applies. And with VSX Systemwide running (but in bypass mode so that no room modelling is applied), the subjective tonal balance of the headphones does indeed change: the upper midrange balance is restored and the slight nasal quality is effectively suppressed. They sound much closer to my K371 reference, and actually pretty good.

To illustrate the comparison between the VSX headphones (both before and after the VSX EQ) and the AKG K371s, I pressed my sound designer wife's Neumann KU100 binaural head into measuring duties and fired up FuzzMeasure. While the KU100 isn't a calibrated headphone measuring rig, it's perfectly able to illustrate headphone comparisons. So, Diagram 1 illustrates a frequency response comparison between the un-EQ'd VSX (red) and my AKG K371 (blue). The most significant differences are that the K371 has more low bass (the K371 is said to comply with the 'Harman Curve' headphone response target, which includes a significant bass lift), a flatter midrange and more energy in the 2-4 kHz region. These characteristics confirm what I heard. I wouldn't read much into the deeper dips of the curves above 4kHz, because the high-frequency region is very much affected by how different headphone ear pads interact with the pinnae, and the generic pinna shape of the KU100 will appear to work better with some headphones than others. Our ears average things very effectively at high frequencies, too.

Diagram 2 shows a comparison between the VSX headphones without EQ, and their response when driven via the Systemwide app. Again, the comparison reflects things much as I heard them. With EQ applied, the VSX headphone response flattens nicely, and it sounds that way too.

So, considered in passive mode I'd put the VSX headphones in the 'competitive if unremarkable' category, and when the Systemwide EQ is added, I'd bump that up to 'really pretty good'. However, the VSX system as a whole isn't just about the headphones, but whether the complete VSX package means an end to mix translation worries.

Software

Rather than use the VSX as a master bus plug-in, I began by firing up the Systemwide app and listening to some familiar material while experimenting with different room/speaker/headphone models. My VSX review sample included the full Platinum set of rooms so I had a lot to play with.

The VSX plug-in and the Systemwide app look the same. In the bottom-left corner are a bypass button and level control. Having a bypass function separate from that likely to be found in a DAW plug-in instance is important, because it enables the VSX binaural room modelling to be switched off without loss of the headphone EQ. Remember, though, that if you're swapping between VSX headphone and loudspeaker monitoring you will need to bypass the VSX plug-in completely, otherwise your monitor will potentially have VSX headphone EQ applied. In this scenario it probably makes sense to have two monitor mix busses set up in the DAW — one for VSX and one for speaker monitoring and mix bouncing.

To the right of the bypass switch is a Depth knob. Depth adjusts the intensity of the binaural modelling, and in doing so, effectively changes the apparent listening position, with more depth making it seem as though the listener is further from the »

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» monitors. The Depth control is deleted for VXS non-binaural models such as headphones. To the right of the Depth knob are 'push-button' switches that provide options depending on the room model selected. A mix room model, for example, might have two or three pairs of monitors (typically nearfield, mid-field, and far-field), and these switches enable their selection. Finally, on the right is an output level control.

Next up on the VSX are five switches that enable room model favourites to be quickly recalled, an option switch that inserts a two-second 'palette cleanser' silence when switching between room models, and an option to insert a five-band EQ in the output. Before I get on to the room models themselves, right at the top of the display are options for Ear Profile and, in two room models, HRTF. The Ear Profile options are designed to accommodate the natural variation in human ear-canal diameter. This is significant because ear-canal diameter influences subjective tonal balance. Smaller ear canals potentially result in slightly emphasised upper midrange. Finally, on the two new room models, VSX offers two HRTF options, previewing a feature scheduled to be released more widely in the next version of VSX. HRTF stands for Head Related Transfer Function, and describes the effect of different head sizes, shapes and ear positions on binaural hearing. We all have a different HRTF, so any binaural recording or reproduction process relies on the use of an average (this is partly why binaural audio works better for some people than others). Providing HRTF options means that VSX binaural encoding can be adjusted to suit different listeners.

The main part of the VSX screen is taken up by the room model browser, with its pictorial representation of the available rooms and their monitoring. Selecting a room is simply a matter of clicking on its icon. Once a room is selected, the VSX headphone feed straight away reflects the modelled audio of the room and monitoring. And that all brings me on to how well it works...

In The Room

I have to admit to being a little sceptical before I began using VSX, but now, with some experience of how it behaves I've come to believe it works rather well. The binaural room models genuinely do a pretty convincing job of moving the listening perspective to another place with an alternative set of monitors. Close your eyes and it's really not difficult to imagine being in a studio hot seat. I've always had a slight problem with the kind of modelled alternative monitor technology of, for example, Sonarworks or ARC room optimisation, because it really isn't possible accurately or fully to turn one monitor into another simply by manipulation in the frequency domain (without any control of monitor directivity and a whole host of other factors). The VSX approach is much more successful because the full signature of the alternative monitors and room are encoded in the binaural analogue.

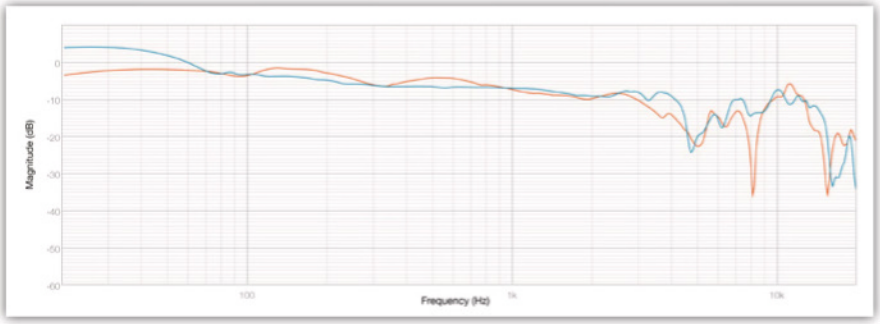

Diagram 1: The frequency response of the Slate VSX headphones without any EQ (red trace), and that of an AKG K371 (blue), as measured on a Neumann KU100 dummy head.

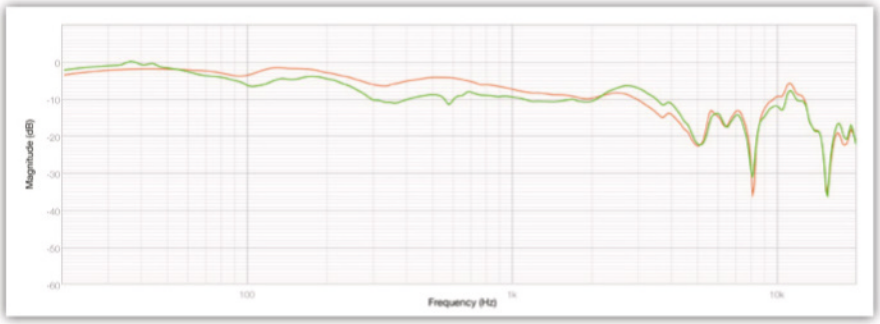

Diagram 2: Comparing the VSX response with and without the Systemwide EQ correction (red and green traces, respectively).

Listening to a range of familiar material and mixes with VSX engaged I found it completely fascinating to hear how things sounded in different rooms. For the most part I was reasonably relieved that music I'd mixed in my own room mostly survived the experience of being played in, say, Steven Slate's room, although there were a couple of "What was I thinking?" moments. The car, boombox and smartphone models too seemed both intuitively convincing and useful also.

I said earlier that I was initially sceptical of VSX, and the reasons for that are twofold. Firstly, how do we know that the VSX room models are accurately representative? They sound convincing, but without access to those specific rooms it's impossible to be certain. And secondly, as with speaker modelling in room-optimisation apps, I can't help but wonder if somehow it's cheating. Traditionally, mix skills are acquired the hard way; by making mistakes and learning from them. And through those experiences, an understanding develops of how to assemble the components of a good mix — one that both does the music creative justice and will translate to other playback systems, with one's own room and monitoring. By offering an easy mouse-click visit to multiple different playback environments, I wonder if VSX is to some extent undermining that learning process. Use VSX and your work will probably translate well, but perhaps you won't have done all the heavy lifting of understanding how and why...

VSX is really convincing, though, so perhaps I'm just old and grumpy. Maybe I should stop worrying about the philosophy, and just get on with enjoying Steven Slate & Co's impressive achievement.

ALTERNATIVES

VSX isn't alone in modelling monitoring environments on headphones. The **Sonarworks SoundID Reference Virtual Monitoring Add On**, **Acustica Audio**'s **Sienna**, **Dsoniq**'s **Realphones**, **Waves' Abbey Road Studio 3** and **Sknote MixingRoom** all aim to do similar things.

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CEntrance Modular Recording Channel

The English Channel

PAUL WHITE

This compact, versatile channel strip boasts SD card recording, USB audio interfacing and mix-minus phone integration.

Chicago-based CEntrance have been building compact, high-quality audio gear for many years now, but their latest offering is a little different from what's come before. Called the English Channel, it's described as a portable, analogue channel strip for recording on location. That might lead you to infer that we're talking about 'just another channel strip', but the modular nature of the English Channel really does set it apart from the crowd.

There are three main elements in this strip: the SoapBox, a combination of mic preamp and dynamics processor; the BlackCab parametric equaliser; and, finally, the PortCaster, which is a USB audio interface and SD card recorder that also offers some online streaming capabilities. CEntrance also make alternative MixerFace and Bouncer output modules that can be swapped into the channel strip.

Overview

Housed in rugged aluminium cases, the three modules are all of a similar size and can be housed together inside the included book-sized desktop cradle, each module being secured to it by a single thumbscrew. A lightweight yet robust plastic transit case is included, which makes this a very convenient system for location audio work. While this is intended as a channel strip, each module is also available as a standalone product (worth knowing if one of them in particular catches your attention while reading this review!) so it's possible to use them separately in the studio if you wish to. For example, you could still make good use of the preamp and EQ facilities even if you weren't using the interface or recorder.

Each module is powered through a USB-C connector, and another USB-C socket allows power to be passed along the channel strip. A suitable compact »

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» power supply and USB cables are provided, along with USB-C to USB-A adaptors should you need them. A small toggle switch is used for power on-off.

The first two modules have balanced XLR combi input and XLR output connectors on their rear panel that allow them to be connected serially. The third module, the Portcaster, has two XLR inputs. All other audio and switch connectors on these modules are 3.5mm mini-jacks, with connection points located on the front edge of each case. The SoapBox, for example, has Smart Link in and out mini-jacks here, along with a line out jack and local/remote switches for the dynamics processing. All the circuitry, other than the interface's A-D conversion, is analogue, and conventional rotary controls are used. This means there's no menu diving or squinting at tiny LCD screens in bright sunlight, and a further benefit is that, unlike some digital systems, analogue circuitry won't suffer from crashes and time-consuming reboots.

Tiny Trio

The dynamics processing built into the SoapBox preamp module comprises a compressor, a noise gate and a de-esser that can be set from 5 to 8kHz. There are high-impedance input options for recording instruments such as guitars, 48V phantom power, and switchable pad and 80Hz high-pass filter controls. Large LED meters occupy the centre of the panel, while further LEDs show activity by any of the dynamic processors, for which there's also a wet/dry mix control. All the top-panel switches, as on all three modules, are recessed and require the included phone-style tool (a toothpick or similar would also suffice) to change their settings, so as to guard against accidental changes when out 'in the field'.

Next up is the BlackCab, a 'British' voiced three-band parametric EQ, augmented by switches for high-pass filter, Air (a spectral enhancer), Pad and Bypass. The EQ bands are LF (72-480 Hz), MF (437Hz to 2.9kHz) and HF (2.4-16 kHz), each with a boost/cut range of ±9dB and a Q range of 0.4 to 8 for each band, so there's plenty of scope here to shape a sound. There's input and output LED metering, too.

Last in line is the PortCaster interface,. CEntrance tell us that the design incorporates their VelvetSound A-D converters and low-noise Jasmine mic preamps, which offer up to 65dB of gain (there's up to 70dB in total on offer in the SoapBox module). This can obviously be used to record into laptops, phones or tablets in the usual way, but even without that, it's possible record audio at 24-bit 48kHz directly to an SD card. What's more, you can record to the SD card and send audio to a connected device simultaneously.

The mic preamp and EQ modules have combi inputs, and the SoapBox can accept instrument signals as well as mic and line levels.

The PortCaster has a two-channel Gain control arrangement for its dual XLR inputs, plus an Aux 3/4 input. Dual optical limiters help avoid overloads. There's also a TRRS input jack that, with a suitable cable, allows you to connect a phone as an alternative source for Channel 2. The connected phone can be set to Mix Minus mode using one of the recessed switches, so that the phone 'hears' the monitor out but with the caller's own voice removed from the mix, which is a welcome faciltiy for podcasts and streaming. The monitoring level can be adjusted from the front panel and there's a control to balance the analogue inputs with a USB feed. A headphone Mix jack is located next to the two XLRs.

The tiny transport/record buttons are located on the front edge of the case, along with the SD card slot, USB C sockets, aux-in and live-out jacks, headphone jack and slide switches for 48V phantom power, Mono/Stereo monitoring and a Lo/Hi (mic/line level) output switch. That obviously makes this part of the system pretty crowded, but the transport buttons are at the top so they're easy to reach. SD cards need to be formatted in the device before use, and should be Micro SD Class 10 or better, with a capacity up to 256GB. The unit is compatible with Android, iOS, macOS and Windows.

All Together Now

To set up the system to handle three microphone inputs, the output jack of the BlackCab EQ needs to be connected to the PortCaster's Aux input. That frees up both PortCaster XLR inputs for use with microphones so that you get one Mic input with access to the Black Cab EQ and all the SoapBox facilities, plus two going directly into the PortCaster. Alternatively, the Aux input, which is stereo, can be used to add music to podcasts and so on. The line output is compatible with DSLR cameras and, for mobile use away from a power source, a suitable USB 5V battery power pack can be used. LED indicators monitor the input levels and limiter activity.

If the system is to be kept tidy, then depending on the configuration you choose you may need to source some short 3.5mm jack cables, but sensibly short XLR and USB C cables are already included. Once set up, connecting the English Channel to my Mac was straightforward, with the PortCaster being recognised straight away as a class-compliant interface. Its headphone output was adequately loud and very clean, and formatting the SD card was a simple matter of powering up while holding down the record button, »

CEntrance
English Channel
$1499

PROS
- Versatile.
- Portable.
- Great sound quality.
- Standalone recording facilities.

CONS
- Compact form means some controls are necessarily fiddly.

SUMMARY
A neat, high-quality and very portable single-channel recording strip with both audio interfacing and SD card recording on board. The English Channel is potentially a great option where mobility is required.

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» then pressing the stop button to confirm that I really meant it. The record LED changes colour during the process, ending up green when formatting is complete. One press of the record button then starts recording, and each time you make a new recording, the result is saved as a separate audio file. Given how much is going on in such a small space, getting the card back out of the unit is inevitably a bit fiddly when you have cables plugged in, so keeping a pair of tweezers handy may be advisable.

I have no negative observations regarding the performance of any of the three modules that make up the English Channel. The mic preamp is clean, and its compressor is well-behaved and works particularly well in conjunction with the wet/dry mix control. Having a variable-frequency de-esser and gate on board is a big plus in situations where some remedial work on the source sound is required — if you're streaming a show live, for example, or want to polish the sound for an online meeting, you can't 'fix it in the mix'. Likewise, the EQ section behaves much like any well-designed three-band parametric console EQ. In other words, it's unremarkable, but in a good way, making it a very useful facility for shaping things on the way in.

Really, though, it's in the PortCaster module that the really clever stuff happens. This works well as a two-channel audio interface, but having two good-quality mic amps on board is a big bonus for anyone who needs the system to mix three mic sources. The phone-friendly Mix Minus feature is a very practical addition, as is the ability to record the proceedings to the SD card at the same time as streaming or recording over USB.

The three modules can be mounted in the supplied lightweight-yet-robust cradle, so that they slope up from the desktop. The cradle is available in a range of brighter colours too...

Verdict

So, what we have here is a well thought-out, compact, high-quality portable channel strip and interface/recorder. An inevitable trade-off of this being a compact and portable solution is that recessed slide switches are used for many of the functions, and 3.5mm jacks are used rather than quarter-inch jacks. I also found some of the text quite difficult to read in subdued light conditions, and had to resort to my LED torch on more than one occasion during the review. Some might not like that the desktop can become festooned with cables, but CEntrance's inclusion of sensibly short XLR link and USB C cables does help mitigate this.

But those minor observations aside, CEntrance have clearly put a lot of thought and engineering expertise into the design of this little system, and they've managed to make it both portable and versatile — more so than might initially be imagined. Importantly, they've also achieved a high standard of audio performance. The included packing case, which also has space for some cables, keeps it safe during transportation, while its compact format makes it convenient for a number of portable audio recording or streaming applications. And while the modules are all available separately, there's a worthwhile cost saving in buying them as a system. ■■■

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Teenage Engineering EP-133 KO II

Sampler & Sequencer

Teenage Engineering's portable workstation offers retro styling and an equally old-school approach to sampling.

SIMON SHERBOURNE

I've often thought that Teenage Engineering's Pocket Operators are deceptively capable little beasts, held back by the size and spec of their hardware. They have a fast workflow and really effective momentary performance effects. The EP-133, or KO II is, in concept, a scaled-up version of the PO-33/KO! micro sampler. It says 'EP Series' on the bottom, so I hope we can expect more grown-up POs in the future.

The EP-133 keeps the basic idea of the KO, providing both drum-machine style one-shot sample playback and chromatic sample pitching. You still have onboard sampling and 'punch in' effects, and both step and real-time sequencing. The KO II remains very portable (potentially still pocketable, in some generous cargo pants) but has massively enhanced capabilities such as full-resolution audio and sampling, four bank layers, unquantised sequencing, velocity-sensitive keys and USB connectivity.

In its new form, the KO II will likely be weighed against other compact workstations like Roland's SP-404 and Novation's Circuit Rhythm. However, my impression is that the key design reference for TE was early Akai MPCs.

Form

Much of the buzz around the announcement of the EP-133 focused on the reasonableness of the price compared to some other TE offerings. Perhaps more significant, though, is what a desirable-looking thing it is. It has a retro style reminiscent of old Casio calculators, hardware MIDI sequencers

Teenage Engineering
EP-133 KO II
$299

PROS
- Immediacy.
- Performance effects.
- The Commit workflow.
- Not limited to four-bar patterns.

CONS
- Only one master effect at a time, and per-Group sends.
- No USB audio.

SUMMARY
Though it's limited in some respects, unlike many sampling workstations the EP-133 gets you quickly to fun and creative places.

or a miniaturised MPC-60. I should say that Retrokits beat TE to it visually with their RK-008 MIDI multitracker, but the KO II has a unique vibe thanks to its display. This pairs a basic three-character read-out with a bank of backlit indicators, maintaining the PO's relationship to handheld games of the '80s: an almost unfair card to play on those of us of a certain age.

As well as the pressure-sensitive mechanical keys (a favourite among gamers and hipster typists) there are three knobs and small slider/fader which are used for all continuous controls and settings across various modes. Some users have reported faulty faders, though it's not clear whether this stems from a component issue or packaging failure. The review unit thankfully showed no issues.

The device really is a lovely thing to pick up and play with, being about the size of a vanilla iPad and surprisingly thin. It's solid, though, with stable rubber feet for tabletop use. Although there's no rechargeable battery, power is handled excellently. Four AAAs keep you working for a good 20 hours, but because power automatically switches to USB when plugged into a computer or charger, I'm still on the first set after a month.

As well as power, the USB port provides MIDI I/O and sample management via a computer, but it can't do audio over USB or direct MIDI hosting. The top edge also sports stereo audio input and output connections, multi-format analogue sync in and out, and TRS-A MIDI ports (adaptors are not included). The audio output doubles for both line out and headphones, and there's also a built-in speaker for casual pickup-and-play sessions.

Function

Each of the 12 dark grey pads hosts a sample, and there are four Groups to play with. The first few projects come pre-loaded with sounds, but you can easily swap any pad's sample by engaging Sound mode and using the +/- buttons to step through the 999 sample slots. You can also type in a number to go directly to a sound. The pre-installed sounds are categorised by hundreds: kicks from 001, snares from 100, and so on. You can also sample directly to any pad, but more on that in a bit.

Each Group can be finger-drummed as a kit, or you can flip a single pad into Keys mode and play it chromatically (or within a scale) from the pads or a MIDI keyboard. Melodic and one-shot mode lanes can coexist happily within each Group, giving you plenty of scope to build up layers, although voice stealing will kick in when you max out the 12-note polyphony. Sequence recording can be punched in momentarily by holding Record, or latched with Record and Play. Sequence length defaults to one bar, but can be adjusted up to 99 bars! While stopped and in Main mode (note the MPC nomenclature) you can step through the sequence in the current Bank with the +/- keys, auditioning as you go. Holding Rec allows you to place hits into steps at your leisure.

Recording is not limited to the grid unless you want it to be. You can Quantise tracks after recording, and even quantise selectively in real time by holding specific pads during playback in Timing Correct (MPC again) mode. Holding the Timing button engages Note Repeat, which can be used in conjunction with varied pressure on the keys to quickly »

The KO II's connections are all found at the top and feature a USB C port and 3.5mm sockets for sync and MIDI I/O, audio input and audio output.

» capture dynamic patterns — we are ticking off my list of must-have features pretty quickly here.

Commit

Everything going on sequence-wise within a Group is lumped together as a Pattern, and you can flip between 99 different Patterns in each Group. A snapshot of which Pattern is currently active in each of the four Groups is called a Scene. Foundational to the EP-133 workflow is the Commit operation, which instantly duplicates the current Scene and moves you into it. This is exactly how I like to work on Push or Maschine, but on a modern MPC it's a fiddly, multi-step process, so I'm glad to see Teenage Engineering going their own way here. You can quickly build an idea, then create variations and song sections without stopping.

Structure-wise, then, you have Patterns in Groups, Pattern combos in Scenes, and Scenes (again up to 99) in Projects (up to 10). By default, flipping between Patterns and Scenes is instantaneous, picking up at the same step in the bar. Again, this is absolutely my preferred workflow, making it easy to perform a fluid arrangement. The only slight hitch is that having to type in a two-digit number to select a Scene slows you down a bit. However, you can use the +/- buttons to advance quickly through adjacent Scenes.

A gaping hole in the functionality at launch is any way to back up or load your Projects and Scenes. When you've filled the slots, you'll need to delete something to start any new ideas. There may be something healthy in the idea of finishing something and moving on, but if you're a live performer, you'll want to be able to load up tunes. Thankfully TE say that a solution for backing up and restoring projects is just round the corner, as well as a way to reload the factory sample content.

Sampling

The EP-133 can sample from the line input or a built-in mic. Tap the Sample button, adjust input gain, then hold down a pad to sample; it's a fast and immediate process, but if you need both hands to produce your sound, you'll struggle. The Sound Edit mode provides top and tail trimming, pitch mapping, an attack and decay envelope, and a choice between one-shot (triggered), key (gated) or legato play modes.

Sample management is handled slickly via a web tool without interrupting other work on your KO II.

You can also time-stretch samples, either relative to a tempo or to fit to a bar length. The time-stretching quality is a bit ropey depending on what you use it for, but it's a really welcome option for working with loops. One thing you can't do is sample during playback. It would have been great to be able to grab phrases and loops in time with the Scene. You can't resample within the device, either. Resampling has never been part of my own approach, but I know it's important to many on devices like the SP-404.

Sample chopping is available, using either the MPC 'lazy chop' method of dropping slice points on the fly, or automatic division. Either way, chop points can be tweaked afterwards. A chopped sample takes over a whole Group, with a slice on each pad ready for resequencing. If you do a tempo match before chopping, all the slices will be in time, which is great for jungle breaks. Another fun jungle trick is to use the Bar match option on a drum loop, then put it into Keys mode. This lets you play the loop at different pitches while maintaining the tempo, with a distinctly Akai S-series grainy stretch. Marvellous.

Sample management and OS updates are handled directly from your computer via web tools, negating the need to fiddle with SD cards or special modes. The EP Sample Tools auto-connects to your device and lists all the sample slots. You can drop samples from your computer into the slots, and can assign samples to pads by dragging to a representation of your KO II. A graphical editor syncs with the hardware, which, as well as facilitating mouse-based editing, provides an enhanced display that responds to the device in real time. This made me think of the OP-Z, which takes advantage of iOS devices as an extended display, but the EP editor doesn't work on mobile devices.

Effects & Fader Modes

A single master effect slot is available, with multiple modes to choose from including reverb, chorus, distortion and

delay, each with two parameters tweaked from the two knobs. The delay is particularly fun. Each Group can be sent to the master FX by degrees using the fader. This scheme works well for performing, allowing you to easily divide a song into basic food groups (drums, synth and so on) and apply effects to them with broad brush strokes. However, the lack of a way to individually treat sounds with effects is one of the KO II's major limitations.

This extends to all the other fader-controlled parameters that you can see labelled above the trigger keys, such as level, pitch and filters. Each of these can be assigned to the fader, with one assignment per Group. This is not as restrictive as it sounds, as every sound also gets its own individual controls for level, pan, pitch and envelope in Sound mode. Groups having to share one filter and send is the main restriction, although for me, the lack of any filter envelope is also a shame (the Circuit Rhythm also lacks this, incidentally). On the plus side, fader movements can be automated and layered.

True to its Pocket Operator roots, the KO II has a suite of 12 performance or 'punch-in' effects. These are momentary effects that mangle the whole performance in various ways. By holding the FX button and pressing the main keys you can apply filters and bit crushing, stutters and loops, slow-downs and pitch warps, and so on. It's not all audio effects: some of the punch-ins scramble sequence pitches, or which samples are played.

The coolest thing is that the effects are controlled by pressure, so filters can be swept, stutter times adjusted and so on by varying how firmly you push the buttons — and you can hold multiple effect buttons at the same time. While the FX button is held, the Group buttons operate as momentary (and stackable) solos and the fader continues to do whatever it's doing, so you can perform endless interesting breaks, fills and builds. A bonus is the Loop mode, which grabs and loops the output while the two knobs adjust start point and length in real time.

It's A Knockout

The EP-133 is super-cute, and affordable by Teenage Engineering standards, but does it have the functionality to compete with other portable sample workstations like the SP-404 or Circuit Rhythm? Yes and no. It has more voices than the Circuit, but lacks that unit's depth of sequencing and arranging abilities. It is outperformed in pretty much all raw specs by the SP, and has a fraction of the others' memory.

And yet... having spent a lot of time with all these devices, I have a hunch this is the one I'd most often come back to. Once you've learned the basics, it's just so quick to create on, and it lends itself to jamming out ideas through organic play rather than fine programming. The sampling workflow is fun and old-school, a throwback to the wonderful Casio SK1 that brought sampling to the masses. You have a core early MPC/SP toolset ideal for Madlib/Dilla sample-chopped hip-hop, or a performance groovebox for improvising a whole house set. I'm looking forward to seeing what's next to the EP party.

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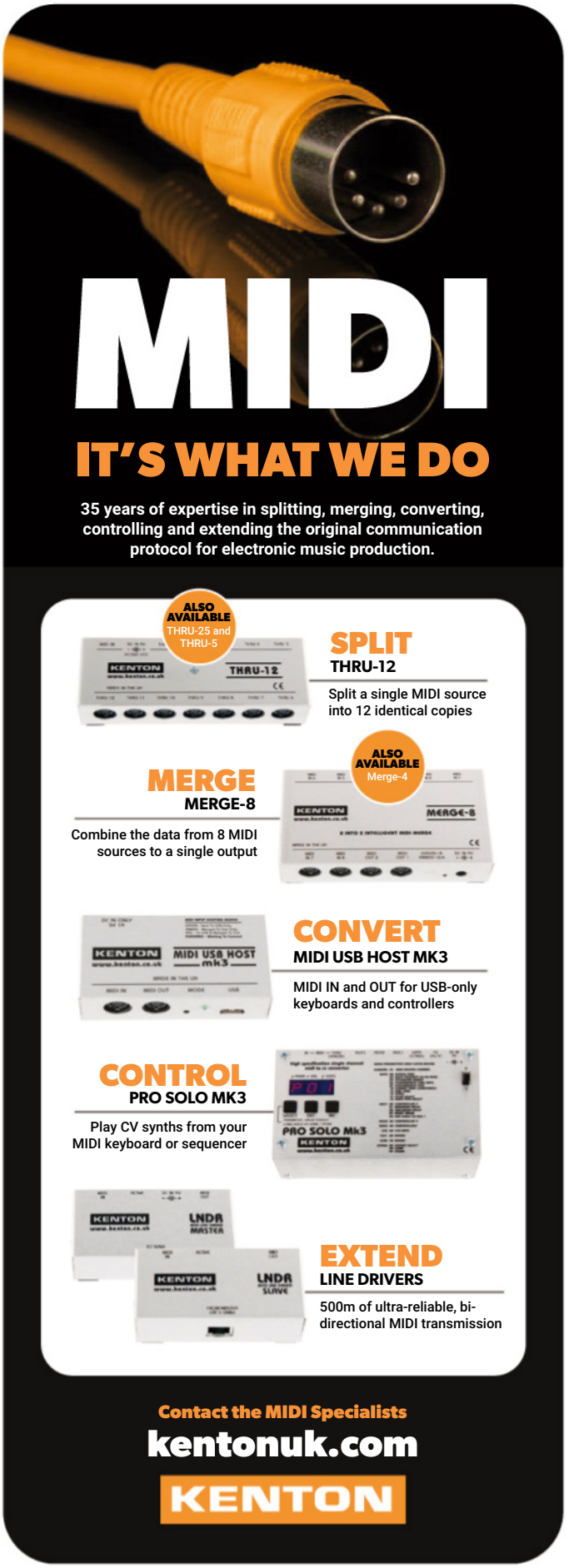

AEA TRP3 & RPQ3

Dual-channel Microphone Preamplifiers

Designed to get the very best out of passive ribbon mics, AEA's preamps offer way more gain than most. But they're not *just* for ribbons...

HUGH ROBJOHNS

AEA's original TRP, which I reviewed it in April 2007 (https://sosm.ag/aea-r92-and-trp), was a compact (half-rack width), no-frills, two-channel preamp with an external power supply. Designed by Fred Forsell, it was entirely dedicated to getting the best possible signal from vintage ribbon mics, which are notorious for their low sensitivity. Consequently, it didn't provide phantom power and featured a DC-coupled input with an exceptionally high input impedance (18kΩ) to minimise the electrical load on ribbon mic motors. Despite the absence of (free-gain) input and output transformers, the active circuitry provided a whopping 83dB of gain using a Grayhill rotary switch (6-63 dB in 12 steps) plus a continuous output level control offering up to 20dB of extra gain (as well as fading down to silence).

Other features included on each channel were a second-order (12dB/oct) 100Hz high-pass filter, polarity inversion and simple traffic-light metering. Internally, the circuitry was based around a combination of discrete JFET gain stages at the front end and op-amps for the filtering and output sections, fabricated using mostly SMD (surface mount) technology, with sealed relays for switching functions. Despite the enormous gain on offer, the design maintained a huge bandwidth (-3dB at 6Hz and 300kHz) for a phenomenal transient response, with a generous headroom margin (+28dBu), and incredibly low noise (EIN figure of -130dBu A-wtd).

So impressed was I with the TRP's sublimely neutral, yet musical sound character, unusually high-gain, and astonishingly quiet noise-floor that I bought the review model without hesitation. I've used it regularly ever since, both with my best low-output moving-coil mics (Beyer M201 and AKG D224E) as well as all manner of vintage and modern mono and stereo passive ribbons. It never disappoints, and always extracts the very best that any of these dynamic mics can deliver.

Of course, being the annoyingly picky soul I am, I identified a few 'cons' of the original TRP. The lack a front-panel power switch was a mild frustration, as was the seven-pin DIN power socket, and the high-pass filter design turned over from too high a frequency to be useful at removing rumble, yet was too steep to properly correct for proximity effects. Not that any of these minor grumbles really bothered me in practice, but I've taken a keen interest in the development of this excellent design since then. We're now on version three of the TRP, which is reviewed here, and there have been a few related products too, not least the RPQ and its own revisions.

Take Two

The first successor to the TRP, unsurprisingly named the TRP2, was introduced in 2018 and although this provided essentially the same functionality and specifications as the original, it benefited significantly from various small improvements to both the hardware and circuitry. The most obvious of these were on the back panel, where the DIN power socket was replaced with a robust five-pin XLR, and the duplicate unbalanced preamp outputs were omitted (the TRP2's balanced outputs could still be used with unbalanced destinations through appropriate cable wiring). On the front panel, the original black stubby plastic knobs were replaced with more elegant long-necked, shiny aluminium ones, and a power button was added as well. So, although the high-pass filter design remained unchanged, two of my petty gripes were vanquished!

On the technical side, the input impedance was raised to 63kΩ and the maximum gain increased slightly to 85dB,

with 7-63dB available on the gain switch while the output level control's range was altered to span -55 to +22 dB. Metering LED levels were also altered slightly, with green coming on at a more helpful -20dBu (instead of -5dBu) and the red illuminating (less sensibly) at +24dBu instead of +20dBu.

Interestingly, the TRP2's specifications claimed that the preamp's bandwidth was altered to -3dB at 1Hz and over 100kHz, instead of the 6Hz and 300kHz of the original.

Undeniably, though, far and away the most significant difference between the original TRP and TRP2 was the addition of phantom power, with individual front-panel buttons (and status LEDs). This rather surprising inclusion was to allow the full gamut of capacitor mics to benefit from the TRP's high gain and low-noise, too. However, adding phantom power inevitably also forced the introduction of DC-blocking capacitors at the inputs, which dismayed some potential customers — so it's worth pointing out that the greater LF extension (-3dB at 1Hz) suggests these capacitors were carefully chosen to minimise the inherent LF phase shift. For vintage ribbon mic owners of a nervous disposition, quivering at the thought of knocking a phantom power button and accidentally destroying their pride and joy, an internal 'no-blow' kill switch was also included to allow phantom power to be disabled completely, if preferred — a thoughtful and reassuring feature.

Another side-effect of adding phantom power was that, when engaged, the input »

AEA TRP3 & RPQ3

PROS

- Stunning performance with astonishing maximum gain yet incredibly low noise.
- Revised high-pass filtering options much more useful than those of earlier models.
- Improved metering and output level control range.

CONS

- It has phantom power and thus inputs are not DC-coupled (unlike the original TRP).
- Removal of 'No-Blow' phantom protection option might concern some.

SUMMARY

The latest iteration of AEA's remarkable TRP dual-channel preamp haS an even more polished and honed feature set, utterly sublime performance, and more gain available than you'd ever want or need!

While the TRP3 has an external switching PSU, the rackmount RPQ3 has a linear one built into the chassis, and connects to mains AC via an IEC inlet. The more fully featured RPQ3 also includes a balanced direct out and line in, the pair of which double up as a pre-EQ insert point.

» impedance inherently falls dramatically — in this case to 10kΩ, purely because of the parallel loading effect of the necessary feed resistors passing +48V into the mic lines. Of course, 10kΩ is still much higher than 'standard' mic preamps (which typically present 1.5-2.5 kΩ), but it's a difference worth noting all the same.

TRP3 Overview

Moving forward five years, AEA have recently revised the design once again to create the TRP3 and the RPQ3, which is based on the same circuit — I'll focus on the TRP3 in the main text and you can find out what more the RPQ3 model brings to the table in the separate box. The TRP3 shares the same format and front-panel layout as the previous models, with some minor aesthetic tweaks and some updates to the core circuitry, which features a discrete JFET front end, a handful of Burr Brown OP1656 dual op-amps for the main gain stages, and THAT 1606 'transformer-like' output drivers.

The maximum gain remains 85dB, with 7-65 dB available on the 12-position Mic Gain rotary switch, and a further 20dB available via the continuous Output Level control. The latter's anti-clockwise position, though, now provides unity gain (0dB) rather than the -55dB of the TRP2 or minus-infinity of the TRP. It's a worthwhile, practical improvement, greatly improving the control's resolution when fine-tuning and matching channel gains. The previous ability to attenuate the output always seemed pointless to me, since such a feature is only really useful in preamps where the front-end is designed to be overloaded intentionally, for musical effect — not part of the TRP's ethos!

Various circuitry tweaks have improved the THD figure by an order of magnitude (from 0.02% to 0.0015%, at 30dB gain and a +4dBu output level). The input impedance has been increased slightly to 68kΩ (falling to 11.3kΩ when phantom is switched on), while other specifications show both the LF and HF bandwidth limits have been raised slightly, reaching -3dB at 10Hz and 200kHz (at the full 85dB gain). Surprisingly, the TRP2's phantom 'No-Blow' switch is absent here — nervous ribbon mic owners beware — but, to be fair, the phantom buttons are stiff enough to resist accidental contact, and the gain knobs long enough to keep fingers well away from the buttons.

A far more significant, pleasing change concerns the high-pass filter: the buttons engaging the 100Hz second-order option in the previous two models have been replaced with three-way toggle switches. These offer two first-order (6dB) slope options instead, with -3dB points at 115Hz or 230Hz (plus off) — a very valuable improvement that addresses the proximity effect issues associated with close miking. The simple trio of level indicating LEDs remains, but the trigger level for the red LED has been reduced to the original TRP's +20dBu level, which more usefully warns of impending converter overload.

Impressions

AEA's third iterations of the TRP and RPQ deliver very nicely optimised feature sets, retaining almost all of the earlier designs' features and qualities, while addressing their very few minor practical foibles. In terms of sound quality, they remain completely beyond reproach in every respect and I rate them as two of the very best neutral microphone preamps currently available. They're blissfully quiet, even at frightening gain levels, and offer a distinctly colourless, neutral yet beautifully transparent, effortlessly dynamic character that's also far from sterile: well balanced, with deep, powerful lows and pristinely detailed transients, yet naturally musical and involving. These preamps extract everything that any moving-coil or ribbon mic can deliver, with impressive clarity and precision. They're the perfect means of accessing the real quality of Coles 4038s, Shure SM7Bs, AEA R44s, AKG D224Es, Royer SF12s... and so on! Highly recommended. ■■■

$ TRP3 $1299. RPQ3 $1799.
T Audio Engineering Associates +1 626 798 9128.
E info@ribbonmics.com
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RPQ3: A TRP3 On Steroids?

For those who prefer rackmount preamps to desktop designs, AEA offer the 1U 19-inch rackmountable RPQ3. Another two-channel design, this is based on the TRP3's circuitry but enhanced with versatile two-band semi-parametric equalisation (though omitting the TRP3's more basic high-pass filter facilities). A distinct feature of the RPQ3 is its inclusion of impedance-balanced preamp direct outputs and selectable balanced line inputs (feeding the EQ section), both of which are presented on quarter-inch TRS sockets. The combination also serves as a practical balanced pre-EQ insert point to each channel. Buttons on the front panel select the line input/insert return mode independently for each channel. The EQ sections are much more traditional compared with those in the very first RPQ (which had a tunable high-pass filter and boosting HF bell section). The RPQ3 EQ offers more conventional, fully tunable LF and HF bell EQs with ±20dB gain ranges (this can be reduced to ±10dB using front-panel buttons). The LF band spans 40-675 Hz, while the HF band covers 2-28 kHz, and both bands feature AEA's bespoke CurveShaper technology, which means the bandwidth (or Q factor) of each bell response varies according to the frequency and gain settings in a musically appropriate way. Further buttons select or bypass each EQ band independently, as well as the entire EQ section. Phantom power and polarity inversion buttons are also present, of course, as is a global power on-off button. The larger 1U rack case of the RPQ3 allows the power supply to be fully integrated rather than external, and this is accessed through a standard IEC mains inlet, switchable from the rear panel for 115 or 230 Volt mains supplies.

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UDO Audio Super Gemini

Polyphonic Synthesizer

If you thought the age of hugely ambitious polysynths was over, think again...

RORY DOW

"20-Voice Dual Layer Polyphonic Binaural Analog-Hybrid with Super Wave Technology" is quite a description. But the Super Gemini has big ambitions. It aims — and mostly succeeds — in standing shoulder to shoulder with synth giants like the Roland Jupiter-8 and the Yamaha CS-80.

UDO are still relatively new to the synthesizer world. In late 2021, they released their first product, the Super 6. Inspired by the Roland Jupiter-6, the Super 6 takes the six-voice polysynth concept to new heights with a true-stereo signal path and FPGA digital oscillators capable of alias-free audio-rate modulation. Like its principal influence, the Super 6 offers a hugely tactile user experience, with physical controls for nearly every parameter.

The Super Gemini takes the Super 6 and doubles it. You get two layers of the Super 6 engine, known as the Upper and Lower layers. True to UDO's desire to keep things tactile, each layer has a complete set of duplicate controls on the front panel. The Upper layer has white fader caps, whilst the Lower layer uses a fetching orange. Of course, that's not the end of it. UDO have added

plenty of additional features to harness the extra power.

It's A Beast

The instant reaction to the Super Gemini is, "Whoa, it's big." At 1040 × 440 × 110mm, it will be one of the larger keyboard synthesizers in any rack. While there are plenty of five-octave keyboards in the synth world, it's the depth that adds extra size, and that's because of the dual set of Super 6 controls for each layer and the new ribbon controller. The keyboard supports poly aftertouch, too, which is a lovely addition.

The overall build quality is superb. It's a beautiful design, whatever angle you look at it from. The white, grey and orange colour scheme is right up my street. I'm also glad to see that the overhanging keys from the Super 6 have gone. They could prove a roadie's nightmare.

Round the back is a nicely recessed panel for the various connections, including stereo mix outputs, stereo upper and lower layer outputs, MIDI In, Out and Thru, a USB Type-B connector for MIDI and file management, and four pedal/CV inputs for sustain, expression, volume and delay freeze (more on that later).

Super Twins

As briefly as we can, let's recap the Super 6, because each layer of the Super Gemini is essentially a Super 6 with tweaks. I reviewed the Super 6 in the December 2020 issue (www.soundonsound.com/reviews/udo-audio-super-6) and won't go into the finer points here, so I refer you to that original review if you want the full details.

The Super 6 synth architecture comprises two 'DDS' (Direct Digital Synthesis) oscillators, two filters, two envelopes, two LFOs, a delay, a chorus, an arpeggiator and a sequencer, all doubled on the Super Gemini. Probably the most significant Super 6 feature is Binaural mode. When engaged, the entire synth works in true stereo, with the signal path — oscillator, filter, amp and effects — having duplicate voices for each stereo channel. In the Super Gemini, each layer can be switched to Binaural mode independently, which is true of most features, as the two layers are independent.

Binaural mode halves the number of voices available, but creates some superb sounds. The Super Gemini can generate 20 mono voices or 10 Binaural voices. So, when using it in Dual or Split mode, you'll get five voices per layer, one less than the Super 6.

Oscillators

The first oscillator — DDS1 — can operate in Super mode, which offers seven unison copies of either standard analogue-type wave shapes or one of 16 single-cycle waveforms. The second oscillator — DDS2 — cannot do unison or single-cycle waveforms, but offers pulse-width modulation on its square waveform and a selection of more normal analogue shapes. It can also double up as an LFO, or as a sub-oscillator phase-locked an octave below oscillator one.

DDS1 is where we find the first of the Super Gemini's audio engine upgrades: wave morphing. You can assign any of the 16 waveforms to Wave A and B, and then morph between them. The Super 6, by contrast, can only play back a single static waveform. Morphing can be automated using LFOs or envelopes, and is available as a modulation destination in the matrix. This is a significant update to the oscillator. It's a shame it's not a full wavetable, but it's welcome nonetheless.

"The Super Gemini is a big, beautiful instrument. Everyone who has entered my studio since it's been here has commented on its good looks, quickly followed by equal admiration for its sound."

A 'DSS Modulator' section offers a wealth of oscillator modulation options, including controls for DSS1's Super mode, dedicated controls for LFO and envelope pitch modulation, and the fantastic Cross Mod, which uses DDS2 to frequency-modulate DDS1. In my original Super 6 review, I did remark that the pitch of DDS1 will move as more cross-modulation is applied, which means that patches could be out of tune with no way to correct them. This is because Cross Mod uses exponential FM, which changes the pitch of the carrier wave. I'm pleased to see two new ways to retune on the Super Gemini. The Lower Layer has a dedicated Detune control to detune the entire layer relative to the Upper Layer, and a Performance preset also has a Detune option. So, if you're doing static cross-modulation, you can retune quite easily.

Other ways to cross-modulate the oscillators are hard sync and ring modulation. The latter is also new to the Super Gemini and is an excellent addition, especially if you want to recreate classic synth sounds. The vast potential when combining cross-mod, sync, unison and ring mod is not to be underestimated.

Filters

The filters are an analogue, low-pass, 24dB-per-octave SSI design based on the classic SSM2044 chip used in the PPG Wave 2.3. They have a lovely resonant character, although some bottom end is lost when adding resonance. This can be mitigated somewhat with the filter drive setting. It's also possible to apply frequency modulation to the filter from DDS2, and it sounds fantastic.

No flagship synth would be complete without a high-pass filter, and it's nice to see that, unlike the simple three-position switch that controls the high-pass filter on the Super 6, the Super Gemini has a slider for full, variable control. It's a shame they didn't make it resonant, but it's definitely an improvement.

Amp & Effects

The amplifier section is reasonably straightforward, with dedicated controls »

UDO Audio Super Gemini

$4495

PROS

- It's a beautiful flagship instrument.
- Two full and independent layers of Super 6 synthesis.
- There are lots of thoughtful additions for sound designers and performers.

CONS

- In Binaural/Dual mode, you only get five voices, not six like the Super 6.

SUMMARY

The Super Gemini is a powerful combo of dual Super 6 synth engines with many improvements like poly aftertouch, ribbon controller, ring modulation, variable high-pass filter, wave morphing, delay freeze input, and more. It's an impressive flagship that can sit proudly amongst the classic synthesizers that inspired it.

» for level, envelope routing, and velocity (just three positions: off, half and full), plus LFO1 and DDS2 modulation. The DDS2 modulation control is new to the Super Gemini and opens up the amplifie to audio-rate modulation, which is an exciting addition. Envelope 2 is the standard amp envelope, but it can be switched to a simple gated mode or a gated mode with a more prolonged release, which frees up the second envelope for other duties.

Each layer has a Juno-style chorus effect, stackable I and II buttons for three different chorus strengths, and a BBD-like delay. I had lots of fun with the two layers of delay. I'm glad UDO didn't skimp by making the effects global.

Modulation

UDO's approach to modulation is to offer plenty of controls on the panel for the everyday stuff. For example, there are sliders for LFO and Envelope control of oscillator pitch, filter frequency and amplitude. There is also a mildly intimidating modulation section above the combined pitch wheel and modulation stick. Here, you can assign LFO2 for vibrato and tremolo. LFO2 and the various aftertouch, portamento and modulation-stick settings are common to both layers, although you can choose to have them affect either or both layers as you wish.

A quick note on the pitch-bend stick, while I mention it. In my review of the Super 6, I expressed some concern about the feel of the pitch stick, particularly when pushing it upwards to apply modulation. It felt very stiff, and was hard to use without the pitch stick flying to the left and applying unintentional pitch-bend. This could have been an anomaly on the unit I was sent, but it was enough of an issue that I mentioned it in the review. The good news is that the Super Gemini pitch stick feels much more responsive, and I haven't experienced the same problems.

For more unusual modulation requirements, each layer has a modulation matrix. Because of the Super Gemini's lack of a screen, this uses the 16 buttons usually used for patch selection. Eight buttons on the left are sources — DDS2, LFO2, envelope 1, velocity, aftertouch, expression pedal, ribbon and key tracking — and the eight on the right are destinations: LFO 1 speed, cross-modulation, wave morph, oscillator mix, high-pass filter, low-pass resonance, envelope decay and delay time. If you don't see your favourite destination listed, don't worry; you can assign to other destinations by wiggling a control. Using the modulation matrix is easy: choose a source and a destination and then adjust the amount knob located next to the 16 buttons.

The modulation matrix is a little different to that of the Super 6. UDO have tweaked the list of available sources and destinations. The obvious example is the addition of the ribbon as a source, and rightfully so. The ribbon is a fantastic source of expression, and its inclusion means you can make the most of it in your sound design. It comes at the expense of the Bend+ and Bend- sources on the Super 6, which is a clever switch. You use the ribbon controller instead of the pitch-bender, which will still work for basic pitch-bending. And because they removed two sources and replaced them with one, it made room for the key-tracking modulation source, which doesn't exist on the Super 6.

The list of destinations has changed, too. Gone are the Envelope 2 release, LFO 1 phase, and delay feedback options, to be replaced by waveform morph, oscillator mix and high-pass filter. These replacements matter far less, because you can always assign one of the eight sources to destinations that are not available on the buttons by wiggling the knob or slider you wish to modulate. So, the envelope release, LFO 1 phase, and delay feedback are still options, even though they're not on the buttons any more.

The ribbon controller and the keyboard's poly aftertouch capability are solid additions. By default, the ribbon controller is assigned to pitch, but when you apply something else, the pitch modulation is removed, allowing you to assign it to as many destinations as you wish. I only wish there was a mode where the amount of modulation would 'stick' when you remove your finger from the ribbon. I found the default behaviour of snapping back to zero when disengaging from the ribbon wasn't always what I wanted. Still, it's a great performance tool.

Double Dip

Dealing with Super Gemini's dual layers is easier than you might imagine. The keyboard can work in three modes: Single, Dual and Split. Single mode plays just one layer, giving you 10 voices in

Arpeggiator & Sequencer

As far as I can tell, the arpeggiator and sequencer are broadly unchanged from the Super 6. Because there are two layers, there is a master tempo clock, and each sequencer or arpeggiator can have a different clock division setting. You can get some nice polyrhythms when layering up two sequenced sounds.

The arpeggiator offers the usual options, playback patterns, octave ranges and swing. The sequencer can store 64 steps, with up to 12 notes per step, and you can program or record ties, rests and accents. The sequencer can also double up as a handy chord memory.

Round the back we find everything you'd hope to find on a large polysynth like this: an IEC mains cable input for the built-in power supply, full-size MIDI In, Out and Thru ports, a USB B port, quarter-inch jack sockets for the four(!) pedal inputs, a stereo mix output and separate stereo outputs for each layer.

Binaural mode or 20 in mono mode. You can use either layer for this mode, switching between them with the layer select buttons. I thought this would be useful for live performances where, for example, you switch between two sounds for the verse and chorus of a song. Sadly, any release stage or delay echo still audible is abruptly cut off when switching, making it less useful in this scenario.

Dual mode stacks the two layers on top of each other. This is your mode for epic layered, detuned, complex patches. It is, in almost all aspects, like having two Super 6s playing at the same time. The Lower layer has a dedicated Detune knob to retune it (±7 semitones) relative to the Upper layer. The 20 voices are split evenly between the two layers, and with Binaural mode enabled, that means five voices per layer.

Lastly, there is Split mode, where you play the Lower layer with your left hand and the Upper layer with your right. You can move the split point anywhere on the keyboard. Like dual mode, the voices are split evenly between the two layers — 10 in mono and five in Binaural mode.

The Super Gemini has two preset types: Patches and Performances. A Patch is a single-layer sound, whereas a Performance saves both layers plus any relevant configuration settings. It's easy to switch between loading either type, and if you're loading Patches, you load into whichever layer is currently selected. I appreciate the simplicity of this. Once a Patch has been loaded into a Performance, it is independent of the original. So you can alter or save over a Patch without the danger of affecting any Performances that the Patch was loaded into.

There are 128 Performance slots and 128 Patch slots to save your sounds. By today's standards, it's a slim number. You won't be buying multiple preset packs from the Internet and storing them all on the machine. You can use the USB connection to mount the internal drive on your computer and transfer sounds as needed, but I feel that would get annoying. Also, because there is no screen on the Super Gemini, there is no patch naming. You have to remember that your favourite sound is in Performance Bank B, slot 4C. This is the one area where I feel UDO could have compromised on their old-school vision. A tiny OLED screen for patch naming would not detract from the hands-on methodology. At the same time, I admire their steadfastness in avoiding that slippery slope.

Pedals

The Super Gemini is generous with its pedal inputs. There are four inputs on the rear: sustain (with auto polarity detection), expression (TRS), sustain (TRS) and delay freeze. The expression pedal is freely assignable in the modulation matrix, and the volume and sustain pedals work as you'd expect and are assignable to either layer. You can even use a dual sustain pedal to control the two layers independently.

The delay freeze input is fun. Connect a single or dual footswitch, and when engaged, the delay feedback will increase to 100 percent while the delay send is reduced to zero. This causes the delay buffer to loop endlessly and »

» allows you to play over the loop. This is commonly known as a 'sound-on-sound' loop: a fine name if ever I heard one!

Conclusion

Let's talk about the most crucial aspect of any synth: its sound. In my review of the Super 6, I summarised its character as "classy". The Super Gemini is no different. The sound matches the build quality. It's refined, spacious, detailed, rarely harsh (unless deliberately so) and highly likeable. The Super Gemini can easily conjure Vangelis-style space leads, faux electric pianos, ambient pads, '50s sci-fi soundtracks, Juno basses, '90s rave stabs, classic brass pads, or '80s pop synths. I could go on. What's impressive is just how well it does any of these. Close your eyes, and you could be listening to a Juno-60, a CS-80, a Jupiter-8 or any classic analogue polysynth.

Whereas the Super 6 was inspired by the Roland Jupiter 6, the Super Gemini feels like a descendant of the Jupiter 8, with a bit of Yamaha CS-80 thrown in. Regarding the sound, I won't compare it directly to old synths because the Super Gemini has its own thing going on. Still, it shares the refined, sometimes grandiose power those machines are famous for.

So, the Super Gemini is an evolution of the Super 6. The base character is very similar. There's just more of it. I suspect many people reading this review will wonder which is right for them. I don't think the little extras like ring modulation, variable high-pass filtering, wave morphing or dedicated delay freeze input would sway anyone to pay extra for the Super Gemini. But the dual layers, poly aftertouch and ribbon controller might.

Having two Super 6 layers at your disposal feels luxurious. There are many ways to use the extra power that the Super Gemini bestows. You might craft a beautiful pad sound on the upper layer, then use the lower layer to sprinkle over a subtle arpeggio or noise that pans around the stereo spectrum. Or, you might use one layer in Single mode to get extra polyphony (10 voices in Binaural mode or 20 with it disabled). Or sequence the Super Gemini from a DAW, for two independent synth sounds with separate outboard processing. Indeed, for anyone who wishes they had a second Super 6, the Super Gemini is an ideal solution.

MIDI & MPE

The Super Gemini is a bi-timbral instrument, meaning that you can play both layers via separate MIDI channels. UDO's implementation is slightly unusual, because you cannot freely set MIDI channels for each layer. Instead, you set one global MIDI channel, from 1-15, which controls the Upper layer, and the Lower layer is automatically assigned to the MIDI channel above that. That's fine, but it does mean that you cannot simulate Dual or Split modes on a single MIDI channel.

The Super Gemini's keyboard outputs MIDI in the same way. For example, if you are playing a Performance preset set to Dual mode (both layers stacked), the keyboard will output on both MIDI channels 1 and 2 at the same time. If you use pitch-bend during your performance, it will be sent on both channels. Depending on your DAW, this could cause problems because some DAWs don't deal with multi-channel MIDI very well (Ableton, I'm looking at you). Of course, that isn't UDO's fault, but it's something to be aware of if you want to record your Super Gemini's MIDI output.

Speaking of multi-channel MIDI, the UDO website says that the Super Gemini is MPE-compatible. However, that functionality is absent in the v1.12 firmware I reviewed. The manual says the MPE button is "reserved for future use". The Super 6 also had to wait some time after release for MPE compatibility and eventually received it, so I don't doubt UDO's commitment. But, if this is an important feature for you, you might want to wait and see how it's implemented.

The Super Gemini is big, but what do you expect with two of everything plus a ribbon controller, arpeggiator, sequencer, and so on? It measures 1040 x 440 x 110mm and weighs in at a healthy 14.5kg.

One of UDO's top priorities is clearly the playability of their instruments. In this endeavour, the Super Gemini does not disappoint. Poly aftertouch, the ribbon, the delay freeze pedal input and the ability to split layers on the keyboard add a wealth of performance options the Super 6 cannot do. If you are a keyboard player, first and foremost, the Super Gemini should be high on your list.

My only real disappointment with the Super Gemini is the decision to reduce the number of voices from the Super 6. "But 20 voices is more than 12!" I hear you protest. And you're right, of course. But two Super 6s would give you 24 voices in total, or 12 in Binaural mode. The Super Gemini has 20 voices, or 10 in Binaural mode. So, if you use Dual mode and Binaural mode together, you are limited to five-note chords, which feels a bit limited. If we're being picky, the Super Gemini is two 'Super 5s'. But, in reality, it rarely matters. Almost every analogue polysynth has voice trade-offs, and the Super Gemini is more flexible than most. Binaural mode isn't always needed; without it, you double that five-voice polyphony, which is enough for almost any combination of sound and playing style.

The Super Gemini is a big, beautiful instrument. Everyone who has entered my studio since it's been here has commented on its good looks and admired its sound. It's a treat to look at and a treat to play. The keybed, the new ribbon, and the consistent tactile feel of every knob and slider all add up to a very classy 'under the fingers' experience.

The Super Gemini is an impressive addition to UDO's line-up and is far more than just two Super 6s sandwiched together. It's always a pleasure when manufacturers release a big flagship instrument. From a business perspective, it's a brave and financially risky thing to do. I wish UDO every success with the Super Gemini. It deserves to do well.

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Abacus C-Box Series Active Monitors

Can these small two-way monitors really deliver accurate sub-bass information at the mix?

MIKE SENIOR

There's a widespread belief among project-studio owners that genuinely insightful sub-50Hz monitoring is simply beyond the capabilities of any affordable two-way nearfield design. Yet, in principle, even a small woofer can generate those kinds of sub-bass frequencies — it's just that you won't get much listening level before the driver reaches its excursion limits and distortion creeps in. In response to this inherent volume cap, pretty much all manufacturers in this space now design their speaker cabinets to resonate at low frequencies, thereby significantly boosting the low-frequency acoustic output before the woofer-cone excursions start maxing out.

The resonance is usually created using a frequency-tuned reflex port or passive radiator, both of which involve some unwelcome sonic trade-offs. Typically the bass level of resonant two-way nearfields falls off rapidly below 50Hz, such that the lowest octave all but disappears and you'll struggle to judge the relative balance of low-frequency components either side of the fall-off point. Low-frequency time-smearing is another a common problem, making kick transients sluggish and generally smudging low-frequency instrument layers together so that it's tough to distinguish between them.

But what if a manufacturer decided not to accept these trade-offs, and pursued superior low-frequency accuracy instead, at the expense of sheer output welly? Well, that's exactly what the German company Abacus Electronics have been doing for years, and it's their current C-Box active nearfield range that's the subject of this review.

Home On The Range

There are three speakers in the range. The C-Box 3 and C-Box 4 are two-way closed-box designs, which have similar phase-plug tweeters but woofers of different diameters (10cm and 14cm respectively). Despite their diminutive dimensions, the loudspeakers boast low-frequency extension down to 35Hz and 32Hz respectively at the -6dB point, and useful audibility of energy well below that because of the comparatively gentle low-end roll-off characteristics of closed-box cabinets. Where you need higher listening levels, these speakers can be joined by the C-Bass 10, a closed-box subwoofer based around a long-throw 10-inch driver. Simple 2.1 bass management is built into the sub's cabinet, with controls for level and phase, for the crossover's high-pass and low-pass filter frequencies, and for a sub-bass cut EQ that helps compensate for low-frequency 'room gain' in small studios.

There's plenty more technical information on the Abacus website, so I won't bother parroting that here, but there is one aspect of the hardware that studio users definitely need to be aware of: all the C-Series audio connections are on unbalanced RCA phonos. I didn't encounter any problems at all in my own studio tests (and the speakers have very low self-noise too), but I do use a filtered mains supply and I kept all my cables as

Abacus C-Box Series
$1160

PROS
- Excellent mix-balancing and mix-tonality comparisons.
- Clean, detailed, and fast.
- C-Box 3 and C-Box 4 deliver incredible low-frequency performance for small two-way nearfield designs...

CONS
- ...but at the expense of considerably lower playback volume.
- Unbalanced audio I/O on RCA phonos.

SUMMARY
Overall, these are terrific specialist mixing monitors at a very competive price, and the C-Box 4 in particular presents outstanding value for money.

short as possible, so I can't say how much unwanted electromagnetic interference these speakers might pick up under less favourable conditions.

Talking Loud

Let's get one crucial question out of the way first: how loud are these speakers? Well, this depends on the bass content of the mix you're listening to. The worst-case scenario is anything with the kind of powerful sub-50Hz kick/bass fundamentals that most quickly max out the woofer's clean driver excursion — tracks like Arizona Zervas's 'Roxanne', Justin Bieber's 'Boyfriend', or Stormzy's 'Big For Your Boots', say. For the C-Box 4, in practical terms that means keeping the listening volume low enough that you can easily have a conversation over the top without raising your voice. While this feels loud enough for mixing purposes, it won't give you much of a physical bass sensation, so you have to get used to judging low-end balances by what you hear rather than what you feel. Nor will this kind of playback volume impress visiting clients, hype up a band fresh from their first live-room take, or inspire a room full of musical collaborators — all scenarios where monitoring wallop usually pays dividends.

For the smaller C-Box 3, mixing LF-heavy productions is unquestionably a quiet listening experience. This is a speaker that should be within a metre of your head to maximise what you can hear, and you'll want to minimise background noise in your workspace too. Under those conditions it's still just loud enough for professional-level work, in my opinion — but it's right on the cusp!

Whichever C-Boxes you use, though, you need to take care with your monitoring volume to avoid unwanted distortion. Unfortunately, the speakers provide no visual overload indication to guide you in this respect, so there's an element of trial and error involved here. I found that experimenting with sine-wave tones helped me develop a sense of how much clean headroom was on offer at different frequencies, but despite this I regularly found myself pulling down my monitoring volume to check whether some bass harmonic I was hearing was in the mix itself or whether I was just driving the woofer too hard. One useful little dodge, though, is that if you're willing to sacrifice some bass extension (perhaps while mixing your guitar and vocal parts), then you can use the C-Box's onboard high-pass filter to cut away the most headroom-hungry low frequencies, so you can listen to the rest of the spectrum louder.

Naturally, if you supplement either C-Box with the C-Bass 10, that reduces the strain on the C-Box woofer, opening up much louder playback levels. With the C-Box 3, I was still a bit reluctant to push things beyond a fairly moderate level (and with the crossover frequency set quite high, around 100Hz or so), but with the C-Box 4 I felt comfortable turning things up as loud as with my own Blue Sky Pro Desk system — in other words, as loud as I've needed for any mix I've done in the past 20 years!

The C-Box monitors feature built-in high-pass filtering (which can come in handy when you need a little more level at the expense of bass extension), and are fed from unbalanced phono inputs.

4 Reference

But what about the actual sound? Well, let's start with the C-Box 4s on their own, because it's easier to discuss the rest of the range in relation to those. The headline here is that the low-frequency extension is extraordinary for such a small speaker. Subby bass lines that simply drop off the bottom of typical resonant nearfields (things like Justin Bieber's 'Boyfriend', Stormzy's 'Big For Your Boots', and the Pussycat Dolls' 'Takin' Over The World') came through with commendable clarity, as did the challenging bottom-octave kick layers in Michael Jackon's 'Invincible' and the subterranean rumbling underneath tracks like Skunk Anansie's 'Infidelity (Only You)' and Post Malone's 'Circles'.

The evenness of the low-end balance is also a real highlight, with the inconsistent weight of the upright bass parts in Sarah Jarosz's 'Take Me Back' and the Steeldrivers' 'Hanging Around' both mercilessly exposed, for example. Sub-30Hz frequencies are certainly quieter than they should be (so the power differential between the lower and higher fundamentals of that Stormzy track isn't as lopsided as I'd expect, for instance), but they're nonetheless still audible enough to provide a great deal of useful mixing information in terms of the timing, envelope parameters, and relative balance in that zone.

The cleanliness of the bass transmission is another strong point — as long as you keep the monitoring volume within tolerance, of course! Sine-wave synth basses remain as stark and featureless as they should be, and you're informed straightaway about the kick-drum LF distortion on One Direction's 'Drag Me Down', Anderson Paak's 'Lockdown', and even Coldplay's otherwise incredible-sounding 'Magic'. As with many closed-box speakers, the LF time-domain response is very well-controlled too, delivering not just the focused kick-drum impact of David Guetta's 'Don't Leave Me Alone', Zedd & Alessia Cara's 'Stay' or Dua Lipa's 'Don't Start Now' with ease, but also damping the ends of low-frequency events assertively enough to beautifully diffentiate between the complex layered bass confections of Ariana Grande's 'Side »

» To Side' and Christine & the Queens' 'Christine'. That last track in particular was hard to stop listening to, in fact, as I've never heard such stunningly clear low end from a speaker at anything like this price!

Overall Sonics

Tearing myself away from the bass hyperbole for a moment, the rest of spectrum has much to recommend it as well. The tonal character feels slightly forward in the 3-4 kHz zone, but once you've acclimatised to this there's an unhyped naturalness and precision to the sound that I found eminently well suited to mixing work. With most small speakers, there'll be a few of my reference tracks where I'll find myself scratching my head and wondering why they're suddenly sounding unfamiliar. Here, however, everything I threw at the C-Box 4 felt natural and believable — not just (literally!) hundreds of reference tracks, but also a number of active mix projects. The high end manages to be both open and smooth, yet doesn't underplay the excessive sibilance of Madonna's 'Sorry' or the spiky vocal transients of Olivia Rodrigo's 'Vampire' and Alison Krauss's 'Paper Airplane'.

Time-domain fidelity across the board feels no less forensic than it does at the low end, and the listening experience is full of detail. The loose clustering of the percussion in Michael Kiwanuka's 'Home' was exquisitely rendered, for instance, as were the surreptitiously automated effects levels in Sierra Hull's '25 Trips'. But I was especially struck by how well this speaker can resolve and interrogate complex distorted textures full of electric guitars and cymbals (such as the Darkness' 'Growing On Me', say), how quickly it identifies unwanted distortion artefacts, and how brutally it communicates the audio consequences of heavily crushed productions such as Devlin's 'Watchtower', Imagine Dragons' 'Radioactive', or Panic At The Disco's 'High Hopes'. The stereo imaging felt very dependable, and there was decent front-back depth too, although on both cases nothing beyond what I'd typically expect of this speaker's market peers.

The C-Bass 10 is based around a 10-inch woofer.

With such a potent combination of bandwidth, detail, speed and tonal discrimination, this speaker really shines when working with acoustic music styles or any kind of high-stakes vocal production — both applications where I think the slight midrange forwardness plays into your hands as a mix engineer. The nuances of audiophile productions such as *The Goat Rodeo Sessions* or Sheffield Labs' direct-to-stereo orchestral recordings came through wonderfully, for instance, as did the gorgeous vocal textures of Solomon Burke's 'Don't Give Up On Me', Crowded House's 'Four Seasons In One Day', and Norah Jones's 'Sunrise'. It was easy to scrutinise the more intense processing of mainstream chart vocals too, with stand-out vocal productions such as Hailee Steinfeld's 'Starving', Little Big Town's 'Girl Crush', and Alan Walker's 'Faded' leaving me in no doubt about their superiority.

> "I've never heard such stunningly clear low end from a speaker at anything like this price!"

But there's one more vital trump card the C-Box 4 possesses: its enormous balancing power. The level of every instrument, voice and effect. The relative levels of consonants, transients and mechanical noises. The level balances between different frequency ranges. It's this kind of balance discrimination that the C-Box 4 provides in spades. Furthermore, the insightfulness of its tone and balance comparisons between different mixes is invaluable when »

Pricing & Competition

Abacus Electronics sell direct from their Hamburg headquarters. Within Germany, the prices (including VAT and free shipping) are: €1290/pair for the C-Box 4s; €990/pair for the C-Box 3s; and €1490 for the C-Bass 10. Sales to the UK and US don't incur German VAT, but Abacus do surcharge for shipping costs and import duty, which means in practice that the rough cost of getting hold of these speakers in the UK is currently £1180/pair for the C-Box 4s, £900/pair for the C-Box 3s, and £1370 for the C-Bass 10. In the US it's currently around $1560/pair for the C-Box 4s, $1160/pair for the C-Box 3s, and $1960 for the C-Bass 10. For exact and up-to-date UK/US pricing, customers should contact Abacus directly.

As full-range mixing tools, I honestly know of no serious competitors to the C-Box speakers at these prices. Every alternative nearfield I've encountered loses out to them in terms of either low-frequency performance or analytical balancing/referencing power — or both! Mind you, mixing isn't the only use for studio monitors, so if you absolutely need more playback volume then almost any project-studio nearfield at this price point will easily outgun the C-Boxes in this respect.

Add in the C-Bass subwoofer and things get more nuanced, because a C-Series 2.1 system no longer outperforms similarly priced 2.1 competitors at the low end, but it also no longer loses out appreciably in terms of playback volume. However, Abacus' overall time-domain precision, sonic detail and bravura balancing should still earn them a place near the top of any mid-price 2.1 monitoring shortlist. Indeed, I reckon the strongest competition for the C-Series in this market sector is Neumann's KH-series — speakers I've also spent plenty of quality time with in my own studio, and which are currently priced pretty much identically within Europe. For my money, a Neumann KH80/KH750 system has a decisive edge over the C-Box 3/C-Bass 10 combination: balanced connections, robust metal driver covers, flexible DSP phase/EQ/delay options, better depth, a touch more detail, and a generally more appealing listening experience. But when you switch to the larger satellites, that flips the sonic advantage marginally back to the C-series 2.1 system for me, on account of its stellar vocal transmission and incisive balance/tone comparisons. So you may face a knife-edge purchasing decision there unless pricing differentials in your specific territory happen to put a thumb on the scales.

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Overall, then, the C-Box is a mightily capable mixing tool and (deep breath) the best all-round mixing speaker I've yet come across at this price. Now, I realise that's quite a bold claim, but I think the low-end capabilities on their own already leave the vast majority of the competition standing, and once you factor in the top-notch balancing/referencing acuity and the tremendous presentation of vocals and acoustic sounds, I just don't see any serious contenders from a mixing perspective. No product is perfect, of course, and there are certainly louder and more aesthetically 'pleasant' speakers available for similar outlay, but this is currently the most affordable speaker system I'd personally be happy using for my own professional mix work.

C-Box 3 & C-Bass 10

Much of what I've said about the C-Box 4 also applies to its smaller sibling. Certainly the C-Box 3's core balancing capabilities and time-domain definition lose very little ground by comparison. The low-end reach of such a tiny woofer is, if anything, even more remarkable, but nevertheless can't quite match up to the C-Box 4 in this respect, and left me less confident in my mix decisions below 30Hz. Partly this was just the challenge of picking out such frequencies at the speaker's necessarily very low monitoring level, where the equal loudness curves can't work in your favour. In fact, in general the C-Box 3's low playback volume required me to give the mix a good deal more mental focus in order to winkle out details. To be fair, I think the degree of detail's almost as good as with the C-Box 4, but you definitely have to work harder to pick everything out!

In addition, the smaller model's just a slightly less appealing listen: a touch harder-sounding in the midrange, and a little light around 200Hz, giving the tone a hint of 'shoutiness' that takes some getting used to. Again, though, once you've acclimatised, tonal relationships between different instruments in the mix feel very natural, and it delivers similarly solid and dependable tonal comparisons for mix-referencing purposes. So while it loses out somewhat to the C-Box 4, the C-Box 3 is still a phenomenal little mixing speaker on its own merits.

Adding in the C-Bass 10 subwoofer was a treat, loosening the playback volume restrictions and extending the C-Box's clean, nimble low end pretty much straight to the centre of the earth! Any residual doubts I had about balances in the lowest octave when working without the subwoofer were swiftly banished and, with a pair of C-Box 4s and the C-Bass 10, I was soon mixing with the same speed and confidence as on my own Blue Sky system.

The subwoofer provides independent control over the high- and low-pass filters, as well as an additional sub-bass filter, and fully variable phase control between 0 and 180 degrees at 80Hz.

Lord Of The Lows

The C-Box 3 and C-Box 4 trailblaze an exciting alternative approach to small-studio mix monitoring, and I think you'd be daft to overlook them if they're within your budget. In combination with the C-Bass 10, both C-Boxes also create very cost-effective 2.1 systems — although I'd definitely recommend saving the extra money for the C-Box 4s if you can. And, just stepping back for a moment, I think it's brilliant that a smaller speaker manufacturer like Abacus can still challenge the dominance of more established global brands, because it's all of us customers who end up reaping the rewards of that kind of healthy competition.

$ See 'Pricing & Competition' box.
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Guitar Rig Pro 7
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PROS

- A vast array of guitar and bass tones.
- ...but great for more than guitar and bass!
- Some excellent new sound-design options.
- Improved workflow thanks to new Sidebar.
- Decent looper and external control options.

CONS

- May not appeal to some with tone modelling OCD.

SUMMARY

For those looking for a single software solution to provide quality guitar and bass tones in almost any style, Guitar Rig Pro 7 has a lot to recommend it. Oh, and it's an impressive creative sound-design tool for other sound sources to boot.

JOHN WALDEN

Whether you prefer your virtual guitar rigs in software or hardware form, we're now spoiled for choice. But I think that's a good thing! All the leading products have different strengths, but for me, one of Guitar Rig's main selling points has always been its breadth: there might be more focused options if you're chasing tones for particular songs or genres, but if you're looking for a 'do it all' option, capable of creating virtually any style of guitar or bass tone, Guitar Rig is a very well stocked one-stop shop. Guitar Rig Pro 7 takes this even further and, I think it fair to say, extends its applications well beyond the bounds of guitar tones.

Now, unless you've been living under a rock, it won't have escaped your attention that iZotope, Brainworx and Plugin Alliance are all now part of the Native Instruments stable. Thus, as well as being available as a product in its own right and as part of NI Komplete, Guitar Rig 7 also forms part of iZotope's impressive Music Production Suite 6, along with Ozone 11 Advanced and Nectar 4. The software can run either standalone or as a plug-in hosted by your DAW. VST3, AU and AAX plug-in formats are supported on both Windows and macOS. An Intel i5 (or Apple Silicon) processor and Windows 10 or macOS 11 (or later) are required. Of course, for playing through the amps in real time you'll need a system that's capable of running at low latency, but Guitar Rig Pro itself is very efficient in that regard.

Plug In, Rock Out

Guitar Rig was already one of the most comprehensive offerings on the market,

Things Without Strings?

For guitar and bass tones, Guitar Rig Pro 7 continues to tick all the boxes in terms of sonic versatility, and the list of components is now very impressive indeed. But Guitar Rig has long been useful for other things, and what's made more obvious with this release — not least because of the excellent new lo-fi effects I mention in the main text — is just what a fantastic sound-design tool it can be for any sound source. With flexible mono/stereo routing, parameter automation and modulation options to be found under the hood when you dig in, and a suite of really creative effects options, this is a powerful multi-effects processor for any sound source including synths, pianos, bass, vocals and drums; it just happens to be in a guitar rack format.

but this release brings us more. In terms of new amps, cabs and effects, there are four new amps (with matching cabs) and five new stompbox-style effects options. These have NI's machine-learning technology (ICM) under the hood, and NI claim this adds greater depth and realism. The amp models themselves include both Fender and Vox inspired options, the Super Fast 100 (I assume based upon the SLO100) and Bass Rage (I think inspired by the Ampeg Venture), and these new models are a real step up in quality — I'll be interested to see if NI eventually apply the same process across the breadth of Guitar Rig's amp collection. Compared with the earlier Fender and Vox models, the newer versions are a significant improvement, particularly in terms of their feel and response to your playing dynamics. The new bass amp is also impressive — it can do a lot more than just the 'rage' I'd expected.

The same can be said of the new stompbox-style effects, which include a new take on the Skreamer (a Tube Screamer model) called Skreamer Deluxe. The sonic differences are for the better if a little more subtle, but this new pedal also boasts three modes and is more versatile as a result: Chainsaw is a distortion for metal tones; Seattle Fuzz is a great grunge-style fuzz; and the IVP Stomp is a preamp with simple EQ that's excellent for dialling overdrive into a clean amp in a very controlled fashion.

IR loaders are now a regular feature of guitar modelling environments, and in a sensible move NI have added one in v7. It works very well too, allowing you to blend up to four IRs with independent control over level and pan, amongst other things. Guitar Rig already ships with a good selection of IRs from the likes of Bogren Digital, Eminence, Lancaster Audio, cabIR and 3 Sigma Audio, but you can also load your own, whether home-brewed or bought from another company.

Get Creative

The looper feature, which had been present in Guitar Rig 5, went missing in v6. This time around we're treated to the new Loop Machine Pro, which is found in the Tools section of the components. What's more, you can configure hardware control for the looper, so if you have a suitable external MIDI controller it would be possible to use that in the same way you might a hardware pedal. However, with options to set the bar count, sync to your host tempo, and use a count-in, it's also fully functional without the need for external triggering.

"You can configure hardware control for the looper, so if you have a suitable external MIDI controller it would be possible to use that in the same way you might a hardware pedal."

In addition to the usual record and overdub options, you get the ability to export both the mixed loop and the individual loop layers, making Loop Machine Pro a neat scratch-pad for developing new musical ideas. Another useful feature is that if you add the component to your Guitar Rig signal chain, and then clear the signal chain, the looper and the existing recordings automatically remain in place. You can then build a further guitar amp, cab and pedal chain to play back the same loop content, which is great for creativity.

There is yet more on the creative front, though: Guitar Rig Pro 7 includes four new lo-fi-themed rack-style components, called Tape Wobble, Noise Machine, Vintage Vibrato and Kolor. These are really cool, and there's a very good crop of presets that demonstrate the additional sound-design options they open up. The first three of these effects do pretty much what you'd expect (and do so very effectively), while Kolor provides a great selection of saturation, overdrive and distortion options, and with a very analogue-esque sound it can be an inspiring sound-design tool.

GRP7 adds four new amp models, including the Reverb Delight and Super Fast 100 shown here, and all were built using NI's ICM technology.

» Finally, we see some fruit of the link between NI and iZotope: the list of components in the Dynamics section now features a compact version of Ozone's Maximizer. I'm not sure I'd want to track through it, but if you need to ensure your guitar doesn't get lost in a busy mix it can be very effective when mixing.

While Guitar Rig's UI provides a perfectly logical workflow, there are so many features on offer that things can get pretty busy in both the Browser and Rack displays. The new Sidebar view is therefore a very welcome addition — this provides a useful visual overview of your signal chain (including for dual signal paths). It allows for easy navigation within the Rack: just click on a component in the Sidebar and it is automatically selected within the Rack, and you can drag and drop Sidebar components to reorder your signal chain, activate/bypass components or delete components should you wish.

Ride In The Rig?

What are the key features a user generally looks for in a software-based guitar rig? Well, first, it has to sound great. Second, I reckon most users appreciate a logical, easy-to-grasp workflow. Third, for some, and perhaps many users, versatility might be important. And finally, for some users, though perhaps not all, it's important that models emulate all the subtleties of the sound and behaviour of the specific make/model of hardware being emulated.

In the context of this wish list, Guitar Rig Pro 7 makes for an interesting comparison with other products made for broadly the same market. For me at least, it comfortably sails over the bar for the first three criteria: it sounds great; the UI offers a good workflow (and, thanks to the new Sidebar, better than what was offered previously); and it is undoubtedly versatile. When it comes to the question of the accuracy of specific models, NI definitely take that side of things seriously, hence the new ICM technology built into many of the new components in this release. All the amp models capture the essence of their hardware inspirations very well, and the newer models are a clear improvement in this regard. That said, I'm not sure it will entice the 'absolute guitar gearhead', for whom a big part of the attraction is the efforts to match the exact behaviour (both good and bad) found in the original amps and cabs being modelled. Tone junkie guitarists and those working in guitar-led music styles such as rock and metal may be better catered for by more niche, specialised options.

With the new cabinet IR loader, you can blend between up to four cab/speaker IRs.

With its routing options, some impressive new lo-fi effects and a version of iZotope's Maximizer included, Guitar Rig Pro 7 is capable of multi-effects jobs that stray well beyond the realms of guitar and bass!

However, if you just want great guitar tones, ease of use, and sonic versatility from a single software solution, and absolutely faithful models of specific amps are less important to you, Guitar Rig Pro 7 could well prove an excellent choice. Indeed, any songwriter or music producer who includes guitar elements in their work could find a whole world of excellent tone-creation options here: grab your guitar and bass parts via DI, and then shape the sounds you want afterwards to best fit the final mix. And the fact that it can also serve as a very creative sound-design platform for non-guitar/bass sound sources should definitely not be overlooked. Considered in that context, Guitar Rig Pro 7 is an excellent one-stop solution. ■■■

$ $199 (discounted to $149.25 when going to press). Also included in some NI and iZotope bundles.
W www.native-instruments.com

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Years of performing boutique upgrade modifications to audio hardware taught us how to build our own high-quality interfaces from the ground up. Boutique component choices; stellar clocking; and the best headphone amps in the game are just the start of the Revolution offering. Now, the legacy of the Revolution 2x2 has grown... introducing the Revolution 6x6 and the Revolution EXP.

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Anatal Electronics XBay 256

Digitally Controlled Analogue Routing Matrix

Software-controlled ‘patchbays’ are invaluable for recall and routing in hybrid studios. Is this the right one for you?

MATT HOUGHTON

Anatal, a Netherlands-based company formed by designer Dennis Bekkering, are the third manufacturer in recent years to launch a digitally controlled analogue routing matrix with sufficient I/O to make it viable as a replacement for a traditional studio patchbay. Currently, the only direct competition I know of comes from Flock Audio (whose Patch range I reviewed in *SOS* April 2021) and CB Electronics (I reviewed their X-Patch 32 in *SOS* August 2021 and XP-Relay in *SOS* August 2023). Each has a slightly different offering, but at heart the proposition is the same: you hook up your audio interface, analogue outboard gear and perhaps a mixing console to these rackmount devices, and you can then use control software to route analogue signals from any input to any output. Manual patching becomes a thing of the past, and recall quicker and more reliable. As the signal path is all analogue, there’s zero latency too. But Anatal take a slightly different, ‘less is more’ approach, based around what Dennis calls Advanced Matrix Architecture (AMA). You can find more information about this on the Anatal website but, in essence, AMA is a setup in which (as with the competition) the routing is performed by analogue 16x16 crosspoint switching chips. Passive types (with no amplification) are used here, so the only ‘amp stages’ in the signal path are those used to electronically balance the input and output. Critically, the signal passes through many fewer of these chips than in a conventional X-Y grid-like matrix. The key benefits, Dennis suggests, are: a very clean signal, no unwanted gain changes, low heat (so fanless cooling, making the device quiet), fewer components (so fewer potential points of failure), and a relatively lower cost per channel.

Overview

Anatal sent me their XBay 256 for review, and this boasts a whopping 128 inputs and 128 outputs — equivalent to 2.67 typical TT bantam patchbays, and enough for a reasonably well-equipped studio. Dennis also stressed that the modular approach to construction means XBays can have different channel configurations: the inputs and outputs are based on eight-channel boards so, starting at a minimum of eight inputs and outputs, you can add inputs and/or outputs in blocks of eight. You can have more inputs than outputs (or *vice versa*) too. I’ve given the price for the standard models elsewhere in this review, but as the price is based on the number of input and output cards, if your I/O needs are more modest the figure will fall accordingly. Those with obscene collections of gear and the funds to match might also be interested in the larger XBay 512: this offers a dizzying 256 inputs and outputs, which is at least twice that of competitors’ nearest models (though I gather the supply of this model is limited.)

A pair of sturdy handles fixes the 4U front panel to the case and makes the unit easy to manoeuvre when (un)racking. There’s a power on-off switch but no other front-panel controls; everything’s configured by the AOS (Analog Operating

System) app, of which more later. On the back is a vast array of DB25 connectors (32 on the review model), wired to the AES59 (Tascam) standard. The channel numbers are indicated clearly enough for normal lighting conditions, but as they're arranged in two pairs of columns (inputs on the left pair, outputs on the right, viewed from the rear), these aren't ever really in doubt. Also on the rear is a chunky twist-lock connector for the standalone linear power supply; a suitable 2m cable is supplied. The PSU connects to AC mains using a similarly chunky cable, again with a robust twist-lock connector. It's very high-quality stuff, and built to last.

Completing the list of external features are a grounding terminal and, for connection to a computer running the AOS software, a USB B socket. Ethernet is an option on some competitors but not here — yet. USB has a limited range so the main control computer must be nearby, but the story doesn't end there. First, the XBay supports more operating systems than most, including Linux and Raspberry Pi. Second, you can work remotely from, say, your DAW machine or an iPad in the live room, thanks to a browser-based version of the app that communicates bidirectionally with AOS running on the main machine. Dennis tells me he's also considering ways to fit a Raspberry Pi inside the XBay, opening the door to various wireless and other connectivity options in the future.

Appy People

On opening the AOS app, you see several menus and tabs. To get started you must first open the Settings dialogue, then select the middle of three tabs, called Device Library. On this page, you define the gear in your studio: you must enter the name and number of I/O for each device, and can optionally record connector types, specify whether phantom power is supplied, set the text and background colours to be displayed elsewhere, and enter general notes. Depending on the version of AOS, you may see a normalling option too; more on that later.

Next, in the Devices In Use tab, you tell AOS which devices are physically hooked up to which XBay I/O. Only once this page is populated will you see anything in the main routing pages. You can assign the I/O individually or map all of a device's inputs or outputs in one go, in which case they appear sequentially (for example, with outputs 1 through to 8, in that order). In the third tab, Settings, you can tweak global settings, including some handy view options for the Matrix routing page.

Most users of multi-channel audio interfaces will find the Matrix page reassuringly familiar, but there are some interesting touches too. By default, the XBay's inputs (attached devices' outputs) are listed in a column on the left, and the outputs in a row across the top, though you can reverse this. To route from one device to another, you just click in a cell. As you move the cursor around the matrix, the rows and columns are highlighted in the device colours you specified when setting up — a more helpful navigation aid than it might sound when dealing with so many connections, particularly if you zoom out to accommodate lots on screen, making the text small. Other nice touches include a tool-tip that, as you hover over a square, displays the units that would be connected if you were to click, and a warning where clicking would create a feedback loop.

> **"As you move the cursor around the matrix, the rows and columns are highlighted in the device colours you specified when setting up — a more helpful navigation aid than it might sound."**

In the default Matrix view, the selected routing is indicated by lines running along the relevant row and column, like a wiring diagram. This can be particularly helpful when you're using the XBay to mult or sum signals: two parallel lines merging into one make this immediately apparent. If doing a lot of routing, though, it can start to look busy very quickly (as with cables trailing across a traditional patchbay!) so, thankfully, in the settings page you can select a different view for the Matrix where connections are represented instead by coloured dots or squares in the relevant cells. What there isn't, but I'd love, is the ability to change this setting when in the Matrix view.

An alternative approach is to use the Chains page. Here, you can again see existing signal chains and create new routings, but this view displays individual signal chains. Unused/unchanged gear is hidden from view (but available from a menu), making it easier to see what changes you've made from the default (see the The New Normal section below). To create a new chain, hit the Add button, move your cursor to the desired source and navigate to a destination and, optionally, to further connections in the expanding drop-down menu. When you've opened the desired signal chain, just click. It's perhaps not as slick as the equivalent screens in Flock's or CB Electronics' apps but it gets the job done without fuss.

Some other, more experimental and (currently) read-only views can help you visualise the current routing: a Patch view has virtual cables dangling between I/O, while the clearest overview is provided by the Network page. There are various other useful facilities, including a handy snapshot-based undo history, a means of determining which devices are displayed or hidden from view (very helpful when dealing with this many channels in the Matrix), and the ability to zoom in/out when in the Matrix view.

The New Normal?

Anatal's website isn't, currently, the best place to obtain the AOS software. At the time of writing, the latest version available for download there for Mac (my main tests were on an M1 MacBook Pro running macOS 12.4, but I also used an Intel i9 9900k-based Windows 10 PC) is still v8. Having assumed this was the latest stable version, I installed it for my tests, but after identifying some 'quirks' in the »

Anatal Electronics
XBay 256
$6,400

PROS
- Huge number of I/O!
- Clean and quiet, with no unwanted gain.
- Can mult and sum.
- App compatible with multiple OSs.
- Browser-based remote control option.
- More to come!

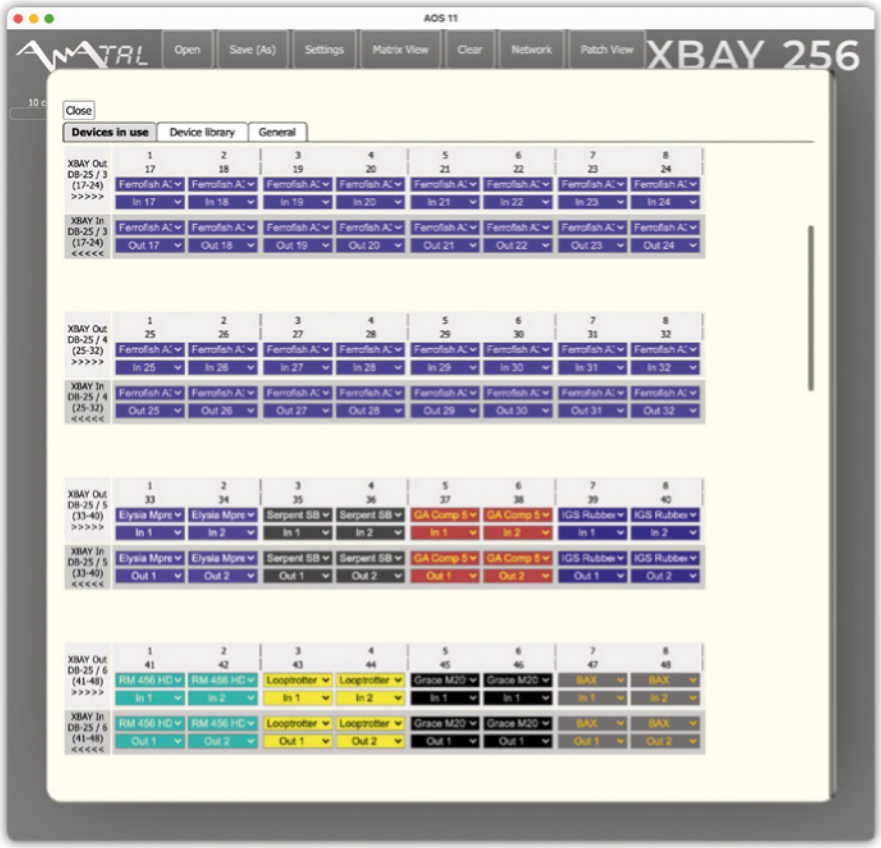
CONS
- The AOS app will benefit from further evolution.

SUMMARY
A capable and high-quality software-controlled analogue routing matrix, the XBay has the potential to replace the traditional patchbay in almost any studio or modular synth setup, and should improve with further development of the software.

» routing (more on that later) I discussed them with Dennis and he informed me they'd already been addressed — and, in fact, it turned out that v11 was ready for release. If you want to keep up to speed with such developments, your best bet is to join the company's Discord server, where Dennis is active and responsive, acting on user suggestions for new AOS features, while ensuring everything's backwardly compatible. Ideally, such details would be published on the company website or at least a forum/site that's searchable using standard browsers (am I the only person who doesn't enjoy Discord?) but it's great that this community exists, and that the manufacturer engages so directly with the user base.

Once I had the right file, it was easy to update to v11, and that was a much more rewarding experience because, alongside new views and other refinements, this gave me a normalling function that was absent in v8. Most people reading this will know what normalling is but, for those still learning, it's simply the 'normal' or default routing configuration: in a traditional patchbay you could set things up so that, with no patch cables inserted, the signal would flow from a preamp to an EQ, then a compressor and thence to your audio interface. Patch in a cord and, depending on the normalling setup, you could re-route the audio, breaking the normal signal flow, or 'sniff' the signal to mult it elsewhere without breaking the normal path.

This was a particularly important feature for me, as it overcame the 'quirks' I mentioned in passing a couple of paragraphs above. Allow me to explain... When using v8, I'd hooked my Ferrofish converters up to the XBay and could see on its meters and those in my RME MADIFace's Totalmix app that when I routed a signal to... let's call it destination A, the signal appeared at the desired destination, but a little of it also appeared at destination B. If I routed a different signal to destination B, that 'unwanted' signal disappeared. It didn't really make a difference in practice, as you're unlikely to be monitoring or recording to an unconnected channel —



Having defined all your studio gear in the Device Library tab, you must then map your Devices to the XBay's physical I/O in the separate Devices In Use tab.

All analogue connections are made using AES59 (Tascam DB25 D-Sub connectors).

I noticed it only because the meters were ‘dancing’ unexpectedly and I happened to be monitoring all my interface’s input channels. Still, it felt rather like a game of whack-a-mole!

A call with Dennis soon resolved the issue: where no active connection is made in AOS, the XBay’s outputs are left unconnected, effectively leaving long, unterminated cables trailing from my interface. All I had to do was make active connections in AOS. When I did that in v8, the Matrix view became cluttered and it was impossible to see the wood for the trees. It also meant that setting a path back to its ‘normal’ routing required more than a single click. So the addition of normalling in v11 was for me a big deal. With the normal paths hidden in the Matrix and Chains pages, things were easier to manage and less confusing. It isn’t yet perfect, as I’ll explain, but note that Dennis plans further changes, based on my suggestions. At present, normalling is set up in the Device Library (where you define your gear; you can revisit this page at any time to make changes). Frustratingly, you define the normalling at the device rather than the channel level, so you must think creatively about defining your multi-channel devices. You may find, for instance, that it’s better to specify an audio interface’s individual channels or channel pairs as separate devices, to match the I/O of specific outboard. Also, it seems you can only create direct connections. I couldn’t set up a mult/parallel path (you *can* create recallable chains with mults elsewhere, just not for the normalled setup).

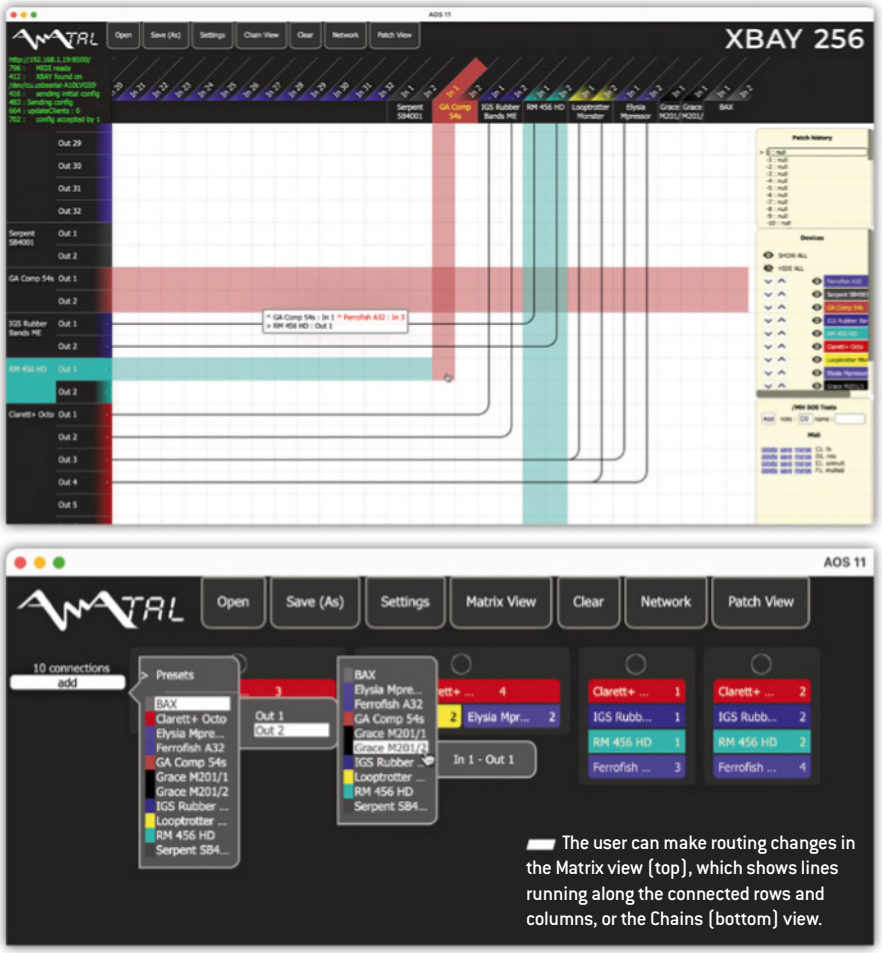
My suggestion was that Dennis create a duplicate Matrix page, with the sole purpose of defining the normal signal paths. These would remain hidden on the main Matrix and Chains pages, as now, to leave those views uncluttered. An alternative might be to add a facility to the existing Matrix or Chains pages to allow the user to ‘write current settings as normalled signal paths’, or some such. Hopefully something like this will be implemented soon, but it’s worth pointing out that, in the meantime, it’s not an insurmountable problem. For example, you can work around all of this to some extent by simply saving and loading different AOS profiles for different projects.

Verdict

At the outset, I discussed replacing a studio’s patchbay with an XBay. Can it do that? Undoubtedly. Had I the funds, I’d have the XBay over a traditional patchbay in a heartbeat, but it has potential in other scenarios with lots of I/O too; a massive modular synth setup springs to mind. The XBay is built to a high standard, and while I didn’t have the opportunity to measure its technical performance, there were no audible issues, and Anatal offer specs and plots if that interests you. Importantly, the XBay didn’t introduce any unwanted level changes, even over some fairly long signal chains. There’s strong competition from Flock and CB Electronics, both of whom offer more features, such as front-panel inputs or controls, and, in the case of CB, the ability to adjust the signal gain at every stage in a chain. But that’s a reflection of the design philosophy, really. There’s much to admire here, too, and I’m sure plenty of people will prefer this ‘less is more’ approach.

The AOS app may be best viewed as a work in progress or permanent beta, but Dennis has been reassuringly quick to act on feedback and embrace ideas, so I expect to see the software mature over time. Already, though, with the latest software it really doesn’t take long to get the hang of things, it already does what needs to be done, and the proposed normalling facility will leave me with a very short wishlist! I love that, as well as running on Mac and Windows, there are versions for Linux and Raspberry Pi, and the browser version too. Not only can you control it from anywhere (in the studio or somewhere else entirely) but you could easily dedicate an inexpensive machine to the XBay, and then update your main DAW machine whenever you like, without having to worry about compatibility problems.

It may be a good chunk of money to spend on something that doesn’t change the sound, but it’s not at all unreasonable: there’s a lot of electronics and R&D time wrapped up in this thing, and it could save a busy commercial facility a lot of time spent on recalls for different engineers or projects — and thus money too!



The user can make routing changes in the Matrix view (top), which shows lines running along the connected rows and columns, or the Chains (bottom) view.

$ XBay 256, as reviewed, $6400. XBay 512 $12,800. Pricing is based on channel count, so XBays with fewer I/O will cost less. Prices exclude taxes and shipping.
E info@anatal.io
W http://www.anatal.io

Heavyocity Gravity 2 Sample Library

Heavyocity's supercharged sequel rockets into orbit.

DAVE STEWART

Seven years after the original Gravity collection achieved lift-off, Heavyocity have launched a sequel. Gravity 2 builds on the format of its predecessor, serving up a fresh collection of textures, risers, swells, impacts and stings along with an intriguing new ingredient: 144 rhythmic pedal loops which can be creatively combined to add impetus and groove to your tracks.

Constructed from over 1000 unique sources, the library features Heavyocity's signature junkyard and mechanical noises, electrical buzz and radio static, digitally mangled acoustic instruments (cello, violin, koto, zither, waterphone, piano) and eccentric performance styles such as bowed oil cans. Also included are a large collection of processed analogue synth signals, and guitar effects created by the company's Neil Goldberg.

This gloriously diverse pandemonium has been crafted into playable instruments comprising both tuned and sound-design elements, thus satisfying the needs of composers who juggle traditional note-based composition with an exploratory sonic approach. Gravity 2 (9.55GB installed) requires Kontakt 7.6.0 or later, and will run on the free Kontakt 7 player.

Overview

Gravity 2's three newly designed Kontakt instruments house hundreds of 'snapshot' presets in themed folders. You can preview sounds in the source browser by clicking on their name: a huge time-saver. The Menu instrument allows you to quickly load sets of 36 sounds assigned to individual keys, while Menu XL's snapshots squeeze 72 sound sources into six-octave presets. The more elaborate Gravity 2 Designer is a three-channel instrument with layered sound combinations designed to spark the imagination and get your creative juices flowing.

As in previous Heavyocity collections, presets can be loaded singly, or in banks of 12 mapped across one keyboard octave. Tuned material is presented in 'low' and 'high' versions with samples mapped according to pitch — pitched samples initially play in the key of C, but you can alter their tuning on the fly with the built-in keyswitches. A handy 'expand source to keys' function maps a selected sound over a full keyboard range for melodic and chordal work.

Rhythmic Pedals

The tempo-synce'd pedal loops add an exciting new dimension to the Gravity experience, covering the spectrum from light percolating pulses to huge-sounding riffs powered by a kickass low end. Multisampled over a wide pitch range, they feature processed analogue and modular synths such as the Moog Minitaur and Sub 37, Lyra 8, Make Noise Strega and DPO, augmented by mutated acoustic sources.

The loops contain much excellent material. I liked 'Cine Sneak', a motoring plucked synth pattern which works equally well for bass parts and ostinato rhythm patterns. In a more aggressive vein, the see-sawing, syncopated 'Zero Day' heavy synth bass riff sounds like a dramatic film cue in search of a movie. Rock fans will also enjoy the funky 'Mechanicals' overdriven palm-muted guitar loop, while the wild octave slides of 'Throwdown' are great for building rhythmic momentum.

Delving into the 'straight high' folder, 'OK Computer' is a rampaging, distorted repeated-note synth pattern that screams to be harnessed to a pounding rhythm track. Other highlights include 'Squealers', which layers backwards swells over a joyfully percussive 16th-note rhythm, the syncopated 'Cuatro Pulsations' acoustic guitar groove, and 'Hammer And String', a lilting, gently insistent processed piano loop which sounds delightful in the upper register.

Three's Company

Bearing in mind that epic film and trailer cues often utilise three-based metres such as 12/8 (or 4/4 with each quarter note divided into three), Heavyocity created half of their 144 pedal loops with a triplet feel. These samples display the same rhythmic drive and diversity as their straight-time bedfellows: 'Chaser' and 'Off The Bottom' are no-nonsense, hustling 12/8 synth bass pulses, and

the danceable ‘Atomic Punching Bag’ exhibits BT levels of programming virtuosity. Higher-pitched presets such as ‘Intelligent Kalimba’, ‘Dream Bells’ and ‘Synaptic Error’ represent the pedal loops’ delicate, ethereal side.

The library’s ‘Rhythmic Moods’ section exploits the Gravity 2 Designer’s three-channel format. The cheerful tick-tock of ‘Clockwork Gravity’ would make a good rhythmic backdrop for an upbeat instrumental piece, while ‘Exotic Escape Plan’ sets a terrific synth bass rhythm against an unhinged stringed instrument upper part. On safer ground, the propulsive ‘Stealth Forces’ sounds like a complete action cue, combining a massive, growling bass with a spiky, twangy processed synth you can use for automated chordal rhythms.

Incidentally, I found many of the triplet-based loops will work in a 4/4 context if you simply count their beat as three groups of four notes rather than four groups of three. This mind game creates the perception of a slower tempo — for example, though it sounds exactly the same, a triplet loop playing at 120bpm would now feel like straight 4/4 time at 90bpm. This mind trick (which drummer Gavin Harrison describes as a ‘metric modulation’) worked particularly well with this library’s ‘Synaptic Error’ synth pulse. I’ll leave you to work out the maths!

Textures

Gravity 2’s sustained looped textures are divided into tonal, atonal and modal categories. The tonal type’s clear single pitch makes them suitable for lead lines, pads and tuned drones, while the atonal sort function more as dissonant sound effects. ‘Modal’ indicates that the samples contain major or minor chord elements, thus making them more musically complete-sounding.

The textures’ moods range from big, menacing cinematic rumbles like ‘Stare At The Sun’ to beautiful pads such as ‘Safe In Your Arms’ (a majestic, gently undulating major seventh chord) and ‘Six String Serenity’, a mystic soundscape with subtle harmonic overtones. Lying in between these two extremes are all manner of mysterious, atmospheric, eerie and dystopian textures perfectly suited to sci-fi, horror, fantasy and psychological drama scores.

Stings

The library’s 252 single-shot stings contain some truly alarming noises: ‘Bass Bender’ sounds like a cross between a detuned fuzz bass and a giant chainsaw, and the industrial-strength ‘Filth Factor’ and ‘Gut Cruncher’ are as terrifying as their names suggest. Neil Goldberg joins the fray with some great guitar slides, scrapes and feedback effects created with an arsenal of pedals (including the aptly named NRG Mauler). »

You can use the large Macro knob to modulate samples’ ADSR envelope, filter, EQ, distortion, rhythm gating, pitch, delay and reverb settings.

The library includes a comprehensive set of master effects.

Heavyocity Gravity 2

$449

PROS

- An excellent collection of textures, stings, transitions and impacts ranging from the beautiful to the devastatingly powerful.
- 144 rhythmic pedal loops provide motor power.
- Constructed from over 1000 diverse sound sources enlivened by Heavyocity’s trademark processing.
- The powerful new Designer instrument offers endless creative possibilities.

CONS

- A library of this depth deserves a proper manual — Heavyocity are working on it at the time of writing.
- Rearranging the step sequencer slices involves a great deal of trial and error.

SUMMARY

If you liked the original Gravity library you’ll love Gravity 2. Comprising over 1000 imaginatively processed sources including found sounds, acoustic instruments, electrical signals, processed analogue synths and guitar effects, it spans the timbral spectrum from subtle to massively aggressive. Its secret weapon is a large set of inspirational rhythmic pedal loops, while the new Designer instrument offers enormous creative opportunities to those who like to dig deep.

» It's not all death and destruction: 'Tracking Signal' and 'Ago In The Future' are charmingly perky synth sounds, while 'Trailer Open Warp' can be transformed into an agreeable marimba multisample by adjusting its sample start time. But this being Heavyocity, the stings are rife with mad, heavily processed, calamitous and iconoclastic noises suggestive of total cosmic annihilation, and sound all the better for it.

Gravity 2's tempo-sync'ed material can be played back at half (0.5), normal or double speed. A dedicated control allows you to adjust the start time of individual samples.

Transitions & Risers

Designed to create an exciting rush into a new musical section, Gravity 2's 'transitions' are two-bar, tempo-sync'ed events which build to an intense climax. The reverses culminate in a reversed-tape backwards whoosh, while the swells reach a peak then subside in a decrescendo. Also included are a set of tension-building risers timed to reach their peak after four bars.

The 144 reverses span an enormously diverse timbral range. Personal favourites include the blasting 'Seven Flare' and the savagely electronic 'Megathon'. I also admired the eerie sci-fi tones of 'Power Surge' and the alien gibbering of 'Venutians', for which Google Translate has so far yielded no results. In the swells department, 'SloMo Fission' and 'Deep Space Messages' are perfect fodder for big-budget sci-fi productions, while 'Ghastly Reveal' and 'Something Wicked' suggest something nasty hiding in the wardrobe. 'Sublimation' also works as a futuristic electronic organ for swelling chord pads.

Gravity 2's risers are considerably more sophisticated than the ascending synth glides of yesteryear. 'Orchestral Cyclone' sounds like a jet fighter taking off as heard from the inside of a tumble dryer (an unusual listening perspective, I grant you). 'The Spins' induces similar feelings of giddiness, while the furious accelerating rotations of 'Cyclone Monster' would make a great intro to a raucous rock tune.

Impacts

Following the format of the original Gravity, Gravity 2 contains an all-new set of 36 impacts containing sub, mid and tail elements which you can load separately or as full mixes. One could describe these pulverising hits simply by replicating the multi-coloured captions that flashed up on screen during fight scenes in the 1966 *Batman* TV series — BIFF! BAM! CRASH! KAPOW! SPLATT! WHAMM! (and so on), but to put it in less comical language, they're brutally explosive.

If pressed, I'd nominate 'Devil's Ringtone' as the ringleader of these crushing impacts — its 'sub' is a huge, epic cinematic drum hit, the 'mid' adds a cataclysmic metallic clang which expires in a horrendous, splintering anguished roar, and the 'tail' section sounds like a malfunctioning circular saw recorded in an aircraft hangar. Nice!

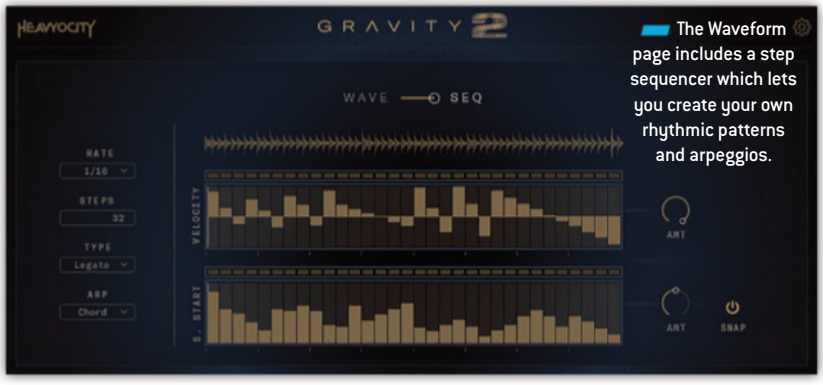
If you want something a little less over the top, some of the sub samples can double as kick drums, to which end you might want to turn off their reverb in the master effects page. The tails can also be used as standalone sound effects. But the main thrust of these hits is Heavyocity's hallmark super-aggressive, overpowering and destructive sonic carnage. To paraphrase TS Eliot, this is the way Gravity 2 ends — not with a whimper, but with a bang.

Conclusion

Well organised, intelligently presented, musically diverse and creatively inspiring, Gravity 2 upholds its predecessor's high standards and ups the ante with some excellent new features. New users will find its easy-to-understand Menu instruments instantly usable, while long-time Heavyocity enthusiasts will enjoy exploring the Gravity 2 Designer. Most importantly, it's a great sample collection with timbres ranging from the beautifully delicate to the crushingly brutal, backed up by an eclectic and exciting set of rhythmic pedal loops to kick-start your compositions.

$ $449
W www.heavyocity.com

Effects & Sequencer



The Waveform page includes a step sequencer which lets you create your own rhythmic patterns and arpeggios.

Gravity 2 is jam-packed with cool programming facilities. Space constraints preclude a detailed examination, but I can reveal that the comprehensive master effects page now includes Heavyocity's legendary Punish knob, a great source of brain-crushing distortion effects. You can use the similarly large Macro knob to modulate samples' ADSR envelope, filter, EQ, distortion, rhythm gating, pitch, delay and reverb settings — this knob can be controlled by the mod wheel, or from the built-in Macro LFO page.

In addition to controlling the start time and playback rate of the samples, the Designer's Waveform page has a step sequencer which lets you create your own rhythmic patterns and arpeggios. It can be used to recompose the rhythm loops' slice points, but in practice doing so is largely a matter of trial and error. That said, I got some great results by carefully adjusting the velocity values and taking pot luck with the sample slices!

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Rode Rodecaster Duo

Audio Production Workstation

Does this new compact Rodecaster achieve the same balance of flexibility, power and ease of use as the larger Pro II?

MATT HOUGHTON

I found the Rodecaster Pro II alluring. Not only does it offer everything you need to create podcasts in a portable package, but it improves considerably on its predecessor in terms of the quality and scope of its facilities. If you've not read my August 2022 review of that device, it may be worth casting an eye over it before reading this one (it's free to read on the *SOS* website: https://sosm.ag/rodecaster-pro-ii). Still, as the Pro II will be overkill for some — not every podcast will have lots of participants, making its channel count, desktop footprint and/or price difficult to justify — it was almost inevitable that Rode would offer a more compact, affordable version...

Cut Down To Size

Launched last Summer, the Rodecaster Duo is very similar to the Rodecaster Pro II: a combination of mixer, multitrack standalone recorder and USB audio interface, and USB streaming device. But it has fewer channels and there are some other subtle differences too. There are four main fader-equipped channels to the Pro II's six, and those faders are shorter than on the Pro II. They're not unduly short, though — ample for the intended application, in fact. What's more, their use has enabled Rode to make the Duo shorter from front to back than its sibling, and it's narrower too, thanks to there being fewer channels: its overall footprint is about 225 x 235mm, while the top of the slanted screen stands about 85mm above the surface on which you sit the device. The smaller confines do mean you're limited to six Smart Pads compared with the Pro II's eight, though. For many users, that will be plenty, but it's something to bear in mind if you're weighing up the pros and cons of both devices. For heavy users of samples, effects and switching things like ducking, it could mean more frequent bank switching. The physical Record button has been replaced by an on-screen button, top-left of the main mixer page, and I can't say I missed it. Importantly, the lovely, crisp colour touchscreen, which is used to access most settings, remains the same generous size as on the 'full fat' version.

As with the Pro II, three rear-panel USB-C ports cater for power (9V 3A; a mains adaptor is included) and simultaneous connection to two devices. These could be, say, a computer for recording and a phone for streaming, but as we were going to press, a firmware

update was announced that, amongst other things, allows these ports on both the Duo and Pro II to host USB mics.

Two conventional mics can be connected, too: you get two of the same excellent mic preamps, accessed through Neutrik Combo XLR sockets on the rear. Next to those are four quarter-inch jacks, providing left and right monitor speaker outputs, and headphone outs for channels 1 and 2. These are the same, capable headphone amps as found on the Pro II, and each has a separate level control top-right of the top panel. A helpful addition is the TRRS mini-jack socket for headphones or a mic/headphones headset on the front (handy, as the cable won't trail across the top). Finally, as with the Rodecaster Pro II, there's an SD card slot for standalone recording, and both WiFi and an RJ45 Ethernet port built in, to allow configuration and firmware updates without having to connect to the Rode Central app running a computer over USB (though I imagine most will opt to do exactly that; see the box for more on the app).

The first two faders are, by default, assigned to the two main mic/line input channels. Using the touchscreen display, you can set these for use with line or mic sources, and adjust the input gain. The second channel can also cater for instrument sources, such as an electric guitar or bass, and again this can be configured at the push of a button and a tap of the screen. As on the Pro II, you can select presets for line, dynamic and capacitor mics, as well as for a number of specific models such as the Shure SM7b, the Electrovoice RE-20 (both popular podcasting/broadcasting mics) and a number of Rode's own mics, including the wireless ones, with which the Rodecaster Duo can pair — a neat touch.

As well as setting a broadly appropriate gain, these presets can include processors and effects, and you can then add/remove effects as you see fit, either using simplified controls intended for non-engineers (such as Depth, Sparkle and Punch) or in an Advanced mode, which gives you access to the more conventional studio processor parameters that lie beneath: a high-pass filter, a noise gate, a de-esser, an EQ, a compressor, an Aural Exciter, and panning. There are also Echo and Reverb effects available here, and yet more effects are available through the Smart Pads.

After the two main input channels, there are faders for two stereo channels: one for the input from a connected Bluetooth device, and the other for the output signal from the Smart Pads. Three further stereo channels that lack physical controls, and whose settings are adjustable only using the touchscreen, cater for inputs from attached USB devices. You can tap on the Bluetooth channel on-screen to set it up, both in terms of pairing the Rodecaster with another device and applying processes and effects to the incoming signal.

Tapping on the Smart Pads channel presents different options: it allows you to configure the pads, each of which can be used to trigger a sample, apply an effect or perform a function. Some functions can mute/attenuate other sounds too (eg. during a censor bleep, 'trash talk', a fade in/out, for back-channel communication such as for a producer to prompt a host, or ducking).

The Smart Pads can alternatively be used to send a MIDI note or CC message, and that opens up all sorts of possibilities, whether for triggering drum machines in your DAW or, with some free intermediary software, switching cameras in OBS. The faders, incidentally, also output MIDI and can be configured as DAW fader controllers. This is handy if you record a podcast's multitracks (whcih can be pre- or post-effects/faders) and want to have hands-on control in post.

For triggering sounds, the pads are nicely configurable, offering latching, momentary and one-shot modes. The effects cover obvious things such as echoes and reverb, as well as more out-there voice changers (robot effects, pitch shifters and the like). These seem to be the same as on the Pro II, so while they sound decent, I won't dwell on them here.

The screen also provides access to various configuration utilities, and these have been made really easy to use. For example, hit the settings 'gearwheel' from the home screen and you can »

Rode Rodecaster Duo
$499

PROS
- Same great preamps as on the Rodecaster Pro II.
- Best 'smart pad' facility out there.
- Supports multiple USB devices.
- Some great DSP effects.

CONS
- None to write home about.

SUMMARY
Every bit as good as the Rodecaster Pro II, this clever device caters for those with more modest channel-count needs and has a much smaller footprint.

The screen rises up from the more gently sloping top panel.

» apply master bus processing (an Aphex Compellor model), map virtual channels to physical controls, toggle channels' solo modes between PFL and AFL, tweak the display settings, assign different colours to the headphone output controls' lights, set up auto mutes for the monitor and Bluetooth output… and more. I covered this in my Pro II review so, again, won't trawl through the detail here. (Likewise, the approach to stereo and multitrack recording, whether as a standalone device with miniature SD card inserted or as a USB audio interface, also remains the same.) Suffice it to say, it's a flexible, accessible and well thought-out system.

Verdict

All in all, I have to say that although the Duo is indeed smaller than the Rodecaster Pro II, it's every bit as good. The mic preamps still sound great (very clean, very low noise) and the same can be said of the headphone amps, which are clean and beefy enough to drive pretty much any headphones. The Smart Pads offer the ability to inject some fun into proceedings through the application of effects and sample triggering, as well as catering for more ambitious setups with ducking, back-channel communication and sending MIDI notes and controller data. There's the usual mix-minus facility where appropriate, too. And if the default configuration isn't to your taste, assigning different sources to the physical faders is drag-and-drop simple.

In fact, for some, this more compact device will be a better choice than the 'full fat' Rodecaster: if you need no more than two mic preamps and headphone amps, and no more than six pads at a time (there are four banks of these available), then it's pretty much a no-brainer. And the addition of USB mic support makes it even more versatile. There's capable competition, of course, from companies like Zoom, Tascam and Yamaha, and it's possible to cobble together a similar system using an audio interface, a DAW and some headphone amps. But I reckon the Rodecasters strike a superb balance between, on the one hand, the accessibility and ease of use required by lay users with little or no traditional audio engineering experience, and, on the other, the degree of control and tweakability that more seasoned producers will crave. With mics and headphones suitably hooked up, someone under 10 could operate this thing: you *can* just insert a card, hit a mic preset and hit Record, and you should get a decent enough result for an amateur podcast or stream. But you can also dive in deeper, configuring processors on the channels and master bus for streaming, while capturing an 'unvarnished' multitrack recording that allows you to create a more professionally polished show in post production — where the Rodecaster Duo can function as a handy controller.

What's more, if you want to do more than podcast, the Rodecaster Duo is sufficiently versatile that it could serve well for camera switching while streaming, or as an audio/MIDI interface for music recording. While it might seem a pricey option if you planned to use it for music production alone, it's not hard to imagine the Rodecaster Duo being used by the same person or studio both for podcasting duties and as the centre of a small home music studio. In short, if you want a hands-on device for podcasting, streaming, and even music-making, it's well worth checking out! ■■■

Rode Central App

As with the Rodecaster Pro II and some other Rode devices, you can do pretty much everything you need using the hardware alone, but you can also use the Rode Central App for Mac and Windows machines to make life easier (Rode Central Mobile, which runs on smartphones, isn't compatible with the Duo). This caters for firmware updates and configuration of the hardware settings, as well as making it easier to manage the Smart Pads and to transfer recordings to your computer, amongst other things. If a firmware update is available, you'll be prompted to install it. I did experience a small quirk with the system: the update from 1.2.1 to 1.2.2 wouldn't install for some reason, so I had instead to perform a factory reset — this took only a couple of minutes, didn't seem to affect user settings I'd changed such as the headphone level knob's LED ring colour, and resulted in the latest firmware being installed anyway.

■ The Smart Pads are not only for triggering samples: they can be configured on-screen or, as shown here, in the app, to engage effects and routing setups, or send MIDI notes and CC data.

■ Like the Rodecaster Pro II — but unlike most devices — the Rodecaster Duo has three USB ports: one for power, and two for simultaneous connection to different USB devices such as a computer and a smartphone.

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MODULAR

Neuzeit Instruments Warp

Eurorack Module

I hate to admit that I am led by my eyes sometimes, since I'm generally a sceptic when it comes to units that threaten to look better than they sound. But when I saw that the rather beautiful light-up 'GalaXY' that occupies the upper third of the Warp's panel is central to its actual operation, I was intrigued.

In one sense it's understandable to call the Warp an oscillator, but in reality it's a self-contained, fully fledged synthesizer — and a four-voice MIDI-controllable and fully MPE-compatible polyphonic one at that, if you add the 4HP WarpEx expander to the equation.

The Warp occupies a relatively modest 24HP of rack space. Alongside the aforementioned GalaXY is a small screen flanked by five encoders, and beyond that a set of seven larger knobs for performative playing. A row of CV inputs and a set of stereo outputs occupy the bottom of the panel. Neuzeit have declared the Warp their flagship module, and for good reason: it is astonishingly deep and incredibly powerful. Based around a clever hybrid engine combining additive and wavetable synthesis, it can generate a potentially infinite range of sounds. Its design encourages impulsive changes with macro controls as much as it does the tweaking of minutiae, all the way down to the last harmonic. We only have a certain amount of column space in this here section, so where to begin?

In the case of the Warp, that would be with the main screen menu. Although the menu system can verge on the laborious, it rarely feels convoluted. In terms of its signal path, that would be with the aforementioned 'GalaXY' of 512 harmonic spectra, arranged in a 32 x 16 grid. Like a conventional wavetable synthesizer, it can move and modulate through complex source waveforms, but it works across two dimensions, with both an X and Y axis (geddit? GalaXY!). As well as sourcing overtones from conventional wavetables, custom harmonic spectra can be generated additively by layering sine waves from scratch. These are then spread across the GalaXY and, crucially, interpolated for smooth transitions between them. Once the GalaXY is populated, PosX and PosY knobs can be used to explore it, with the pickup point represented by a rather lovely floating light — or lights, in polyphonic mode. TL,DR: design waves, drop them on the Warp's GalaXY and then move around it as if doodling with an incredibly pretty Etch-A-Sketch.

Its four voices can be sent to different areas of the GalaXY, meaning that in mono mode it's essentially multitimbral. With the addition of some modulation from the Warp's very deep modulation matrix, it's possible to achieve phenomenal movement and texture. Even deploying the simplest asynchronous LFOs across the X and Y dimensions makes for excellent results.

With the GalaXY charted, it's then a case of using the Warp's Spec, Warp and Detune sections to map performance functions across its parameters — often several at once. Think of these knobs as being designed to counterbalance its incredibly deep and very-nearly-but-not-quite-too-fiddly aspects with broad, performative gestures. And I must say that in this department the Warp's layout is pretty impeccable. I for one very much like the prospect of spending as long as I need in the studio fine-tuning a preset, but also being able to save it with some wild variation potential nestled behind a manageable handful of controls. I was able to flick through distortion, filtering, bit-crushing and more, quickly applying one or all of them to the Warp knob, with a response of my choosing for each. Twist that knob (or apply CV to its input) and hear the entire sound of the module shift

— or disintegrate completely. The Spec knob is likewise loaded with potential, controlling a state-variable spectral filter for intense one-knob tonal sculpting. It can introduce make-up gain to compensate for filtering, and can even add additional overtones to give the source audio a harmonic helping hand if desired.

Whew. I told you the Warp is deep. But it's accessible, too. Keep Calm And Use Presets, joke Neuzeit in the manual. Indeed, if you don't fancy the deep-diving part, its extensive library of fully programmed presets remain ready and waiting for gleeful sonic performance. Original but usable, detailed but performative, I'd call the Warp a category leader — if only there was anything else in its category. *William Stokes*

$ $699

W www.neuzeit-instruments.com

Error Instruments Brinta

Eurorack Module

Welcome to another joyfully peculiar Error Instruments protrusion into normal space. It arrives in the form of a porridge-inspired granular sampler that's as brilliant as it is baffling. At times you feel you are meddling with forces you can't possibly understand, until a random turning of knobs pulls everything into a moment of focus and you realise "Oh, I've been playing with a drum loop."

Brinta is the latest collaboration between Error Instruments and This is Not Rocket Science. It's an exciting creative space where TiNRS get to play with their weirdest algorithms, and Error Instruments get to break them. The result is immensely playful and experimental, inviting exploration rather than analytical inquiry. Which is a long-winded way of saying that I don't quite understand it, but it's a heck of a lot of fun.

The basic idea is that you sample something into the granular engine and then mess about with the cascade of tiny slices or grains of sound that pour out. The action takes place in the 'golden grain circle' that glows invitingly at you from the middle of the module, visualising all sorts of activities. It shows the position of the playback and recording heads; it glows blue with high frequencies, green for mids and red for lows; and it documents the life of little golden playheads that materialise into being for the duration of the grain before vanishing from existence.

Turn the big knob in the middle to set the position of the playhead. It's affected by the Speed control, which moves the playhead through the sample from that position. However, the Pitch control also affects the pitch at the playhead position, by speeding things up and slowing things down. Under certain conditions, the way in which these three functions interact becomes clear, but more often, it's lost in the smush of granular beautifulness. Using the Size knob, you can drag some clarity out from the wash of large grains into the stark abruptness of tiny ones, but you never quite get there.

Finally, we have the X control, which means different things depending on which of the three modes you've selected. In Cloud mode, X deals in density, where you find the more traditional granular effect of shimmer and light. In Chord mode, the grains are shifted in pitch to generate either major or minor chords depending on which way you turn the X knob. Mode three is a harmonic probability function created in honour of Kid Baltan. As you turn the knob to the right, the probability of the pitch doubling again and again increases. All the modes mode are lots of fun to play with, and they are similar enough to switch between without too much of a jerk, while taking you in very different directions.

Brinta comes with five samples ready for your explorations. On my first go, these felt loud and slightly mad. You've no idea what it is you're listening to, and randomly turning knobs does very little to provide any illumination. What you do have is a soundscape of weird and exciting granular stuff that starts to get frustrating because you don't know what it's supposed to sound like. Once you start recording your own samples, though, it all starts opening up, and I found that using my voice was a great way to understand what Brinta was doing. You have a choice of two inputs. The first one feeds the granular engine without any monitoring, whereas the second mixes your input with the output. The output is in stereo, with half the grains going left and half going right. Listening to it in stereo is definitely worthwhile.

Hit record, and a red dot runs around the circle displaying your sample's recording time and length. Press play to get the granular engine working on the sample. You can enable both play and record to use it like a granular delay effect that's constantly overwriting itself. Once you've recorded something recognisable, all the controls begin to make more sense. You can shift the pitch of the playback with the Pitch knob, and adjust the speed of the playhead going through the sample with the Speed knob. It's possible to find the right pitch and speed to play the sample back as it was recorded by watching for a pink flash in the circle as you turn the knobs. You can also halt the movement and use the Position knob to pick out sample slices and watch the golden heads sparkle out from your audio. As you push parameters and turn knobs, you tend to lose all sense of what everything is doing, but I don't think that really matters. What's important is that you are lost in a wonderful place of weird and beautiful occurrences. If you then introduce some sources of modulation, Brinta becomes a mystical engine of improvised soundscapes and occasional hilarity. *Robin Vincent*

$ €250

W www.errorinstruments.com

Modular Profile: Thomas Hutmann

As Neuzeit Instruments, German designer Thomas Hutmann is creating marvellous, original designs for Eurorack. From the Orbit synth voice to the Quasar binaural audio mixer, Neuzeit designs sound excellent, all but throw the rulebook out of the window and look rather gorgeous too.

On his entry Into modular

I never owned a modular system until I started developing my own modules. Before that, I was producing techno with a non-modular synth setup and Ableton, but I also did a lot of circuit tinkering on breadboards and circuit-bending on some Korg Volcas. When I decided to build my own hardware synthesizer, I couldn't find a decent case and power supply anywhere, which is when I discovered Eurorack. It meant certain things were outsourced to the user, and it having patch points was a real feature, as opposed to just having solder pads for circuit-bending that you could only access by voiding your warranty! Later my first module, Orbit, was born. I decided to also let the user choose the oscillator of that synth, so you simply have to add any audio source and get a full synth voice based on bit-crushing and distortion.

On his go-to modules (aside from your own!)

In my big 'fun case' I have a lot of Doepfer modules for everything analogue and some Xaoc effects like the Sarajewo BBD Delay and the Timiszoara Multi-FX. I love their clean, scientific silver look! For the harder side of the spectrum, I am a big fan of Schlappi Engineering modules. Also the Droid series by Mathias Kettner, aka Der Mann Mit Der Maschine, is a great toolset for CV and MIDI tasks. I also do a lot of hardware design for Mathias, so my Droid controllers are all a bit Frankenstein-ish as they are mostly prototypes. I've actually gotten most of my modules by trading, so I only really own modules by like-minded engineers I know personally and with whom I feel I have a relationship.

On Neuzeit Instruments

Neuzeit means 'new age' in German. The products I make are both meant for now and for the future. I do my best to make durable and sustainable gear by using plastic-free packaging, to give it a timeless interface, and provide physical robustness that will last for decades rather than becoming mediocre disposable electronics. To me, a true Neuzeit Instrument is one that offers hands-on expressiveness, but also has its own character, a beautiful interface, and enough feature depth to give the musician years of enjoyment.

On the Warp

Warp is the result of several months of research in which I tried many, many things with partials and additive synthesis. This included training neural networks on audio, building algorithms for polyphonic fundamental analysis and time-stretching, and much more that did not make it into the final module! However, the goal of this journey was to find an approach for deep-level access to harmonics and some sort of semi-automatic generation of a soundpool, which finally resulted in the GalaXY editors. This would have been good enough for an additive oscillator, but I felt it was not yet a real 'instrument', so I gave the Warp the most powerful CPU I could find and squeezed every last bit out of it, eventually adding a fully polyphonic wavetable synthesizer engine!

On the culture of modular

Modular is a great melting pot of sound designers, the DIY community, engineers and musicians, most of whom have been in the game for several years and are dedicated to their passion. It is also a great way to connect with people at synth shows, jams or by trading modules. As a developer, I am also glad that there is an environment where you can pick a specific part of the signal chain and focus on it, instead of having to build everything else as well. It is also great to see that most of the modules out there have a decent build quality, which ensures a vibrant secondhand market and keeps them in the loop for a long time. *William Stokes*

What's New

If you're not familiar with **AI Synthesis**, get to know them: Abe Ingle's Portland, Oregon-based operation is a DIY powerhouse and not long ago unveiled their latest offering. The AI024 X VCF is a four-pole low-pass filter promising to bring the "clean, classic, creamy" character of legendary '80s synths to Eurorack. www.aisynthesis.com

ADDAC System, meanwhile, have announced the launch of the ADDAC309 CV to Expression, a simple but effective 4HP module that allows CV to be routed to the expression input of any effects pedal. The best part is that it draws all its power from the pedal in use, so doesn't need any power from your case. www.addacsystem.com

Bastl Instruments have launched the seemingly Matrix-themed Neo Trinity, an 8HP 'automatable modulation hub' boasting six channels of LFO, Envelope and CV generator-based goodness with incredible amounts of flexibility. www.bastl-instruments.com

ALM have also released a utility 'smorgasbord' recently and also in 8HP. The Mega Milton includes a stereo line input converter for boosting line-level audio, a four-input fixed mixer, a gated slew limiter, a sample & hold with analogue white noise and even a buffered mult. Whew! www.busycircuits.com *William Stokes*

Win! Heritage Audio i73 PRO Edge

Worth $1499

Based in Madrid, Spain, Heritage Audio were founded in 2011 with the aim of making high-quality, vintage-sounding gear available to the masses. They've released a number of products based on such legendary gear as the Fairchild 660 compressor and Motown EQ, as well as their own original designs.

For this exclusive *SOS* competition, we've teamed up with Heritage Audio to offer you the chance to win their brand-new i73 PRO Edge. This premium USB audio interface combines a pair of preamps based on a legendary British circuit, plus a host of features that make it ideal for integrating into modern workflows.

To enter, please visit:
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The preamps employ Class-A transformer-balanced circuitry to impart analogue mojo to all of your recordings, and can provide up to 70dB of gain on mic signals. They offer all the features you could wish for, from stepped gain controls to polarity switching, an input pad and phantom power. Mic and line signals are catered for through combi inputs on the rear panel, while two jacks on the front allow for convenient DI'ing of guitars and basses via a J-FET input stage. A separate output gain trim for each preamp allows you to drive the input transformers while still keeping levels manageable.

The right-hand side of the unit is dominated by a monitor control section, which is based around an endless encoder plus buttons for mono, dim and mute. There are also two independent headphone outputs, whose levels and mix sources can also be controlled using the rotary encoder.

But the i73 PRO Edge is more than just a stereo interface: in addition to its preamp inputs and monitor outs, it boasts a pair of line-only inputs, plus two further line outs — meaning you can use external hardware in your DAW, or hook up a secondary pair of monitors. It also sports optical ADAT ports, making it easy to expand the I/O. Internally, the interface hosts a mixer that caters for latency-free foldback mixes and even analogue-modelled processing, courtesy of a growing range of plug-ins. These include emulations of a classic delay unit, an iconic bass amp, a vintage tape recorder, a plate reverb, and even Heritage's own BritStrip channel.

To be in with a chance of winning this fantastic interface, simply follow the URL shown, and answer the questions there, by Friday 5th April 2024. Good luck!

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Icon Pro Audio V1-M & V1-X

DAW Control Surfaces

Are Icon's latest range of control surfaces the perfect interface between human and DAW?

Icon Pro Audio
V1-M & V1-X
$1099 & $899

PROS

- Well-built, weighty unit that feels like a quality mixer.
- Includes templates for all common DAW packages.
- Rugged rubberised buttons, 100mm faders, full transport control and jog-wheel functionality.
- App-based assignment of faders, pots and buttons.
- Expandable up to 32 channel faders.

CONS

- The footprint is relatively deep; it could be a squeeze on a desktop alongside a computer and music keyboard.
- Plug-in control and navigation isn't as seamless and easy as you might hope.

SUMMARY

The experience of using the V1-M as a primary interface for your DAW is an undoubted pleasure. It undertakes tracking and level-mixing tasks beautifully, becoming less desirable the deeper you go in the mixing process. It's not time to lose your mouse just yet, but it's getting closer.

DAVE GALE

I still recall the day I gave in to the DAW powers-that-be and surrendered my generously proportioned mixing console in favour of mixing in the box. While I have never had a moment's regret on a sonic level, I've always felt like I lost a piece of my soul that day, leaving a void to be replaced by a mouse and a computer keyboard. So began the search for the perfect DAW control surface that could capture the spirit of a console while keeping my mouse at arm's length.

Total Control

Icon Pro Audio are no newbies to the control surface arena. Their QCon series has always looked the part, and they've since added a wide range of other products at different sizes and price levels.

Jumping to the top of the tree, I had the opportunity to explore the flagship V1-M with a V1-X expander. The V1-M is the starting point for anyone wanting a single, high-quality controller device. What you get for your money is something that looks and feels just like a quality audio mixer. It hosts 100mm motorised faders, chunky buttons for activities such as soloing and muting channels, and similarly reassuring transport controls, including a beautifully weighted jog wheel. There are eight channel faders on the main unit plus a ninth master fader, and there's the capacity to add a further three eight channel expanders to take your channel fader count up to 32. If that's a luxury you cannot afford, either in price or desk space, you can easily use a single unit to shuttle around your DAW's track list, guided by a bargraph VU meter and scribble-strip display at the top of the panel.

The Setup

The V1-M is the largest controller of the range, especially with regard to depth measurement. At 380mm deep, it was just about possible to place the unit in front of my computer monitor, with the computer keyboard in front. If you also have a MIDI keyboard on your desk

the depth will likely prove too much for a standard desk from a favoured Swedish furniture shop. Placing the V1-M to one side is an option, but given the potential for mouse-less DAW integration, it feels like you will want to have it right in front of you, which might prompt you to consider the smaller models in the range.

Once in place, the V1-M requires both power and a USB connection. The V1-M has a Type-C port, and a USB A to C cable is supplied, but not a C to C cable.

The V1-X expander can be attached to either side of the V1-M main unit, meaning that you can create a very elegant setup with the transport and master fader located where you want them. You just have to tell your DAW how they are arranged on your desk and the driver does the rest. You will have to have capacity to connect each device to a USB port, though, which may require a substantial powered hub.

Next, you need Icon's iMap software, which is the host software application and runs alongside your DAW. Apart from driving the unit on the software side, iMap also allows a huge degree of personalisation. As the V1 is a universal control surface, you can select an appropriate template for your DAW, with 18 templates available for all the usual suspects and also some less well-known platforms. You can load templates for up to three different DAWs, or different templates for the same DAW, quickly switching between them with the three dedicated DAW buttons located at the top of the control panel.

Like most small hardware controllers, the V1 uses the Mackie Control and HUI protocols to communicate with the DAW. I tested the V1-M with Logic Pro X, entailing the use of Mackie Control. Having selected the appropriate DAW template it was then just a case of visiting the Control Surface page in Logic to initiate the connection within the software. Hey presto: the two were communicating and all was well!

Test Driving

At first sight, the integration at what we could call the primary level is really excellent. Shuttling back and forth, using buttons or the jog wheel, is a very seamless affair and incredibly responsive too. Zooming in and out on DAW windows requires a press of the Zoom button and a rotate of the jog wheel — which, while easy to activate, can be a little over-keen! The jog wheel is weighted, and you can find yourself free-wheeling fairly swiftly.

The iMap software allows the selection of multiple DAWs and reassignment of any button, pot or fader on the unit.

MIDI Reassignments

A V1-M feature which I was keen to explore is the ability to reassign faders to MIDI duties for the control of sample libraries. I'm used to having three MIDI faders at my disposal for controlling orchestral sounds, so having the ability to flip the V1-M to the MIDI side to use in this way, before flipping back again is a pretty fundamental ask.

Using the iMap software, it was easy enough to reassign faders to a MIDI CC operation, but incorporating this into a template alongside DAW control proved more difficult. Even after seeking advice from Icon, where the consensus was that creating a second DAW level on iMap would be the way to go, it still didn't work seamlessly or successfully, which was a great shame.

The iMap software is constantly under revision, and given the newness of these devices there's scope for an easier method of MIDI interaction to follow in the future.

While it's never going to be possible to please all people at all times, I would question the ultimate layout of the transport section. The jog wheel occupies the area to the bottom-right of the main unit, and while this means that it's easy to use in collaboration with the associated buttons that surround it, it does obscure access to the main transport controls for stop and play. I found myself knocking the jog wheel on a regular basis.

Meanwhile, back in the channel zone, adjusting levels using the faders while also selecting, muting and soloing channels is exactly what you would hope for. If you're working in a large project, navigating from one bank of faders to the next is executed with comparative ease, and is aided by the responsive old-style segment VU metering and virtual scribble strip, which reflects whatever track naming you apply within your DAW.

Function Layers

When it comes to controlling and editing other DAW elements, such as instrument or plug-in settings, you're required to head toward the Function layer section. Located above the transport controls are 24 virtual touch-buttons. The 6x4 layout behaves much like a Stream Deck, with five Function Layers. The first three relate to DAW-based operation such as track and plug-in selection, with the remaining two providing global DAW operations, such as open/close project. Given that there are 120 assignable buttons here, it's not surprising that only half of them »

Lining Up The Products

The V1-M and its associated expander are not the only new Icon kids on the block. The P1-M and P1-X are the next models down in range, and effectively offer the same functionality in a footprint half as deep at nearly half the price. They look and feel much the same as the larger siblings, providing a workflow experience which is identical. Apart from a shrinkage in size across the whole unit, the faders remain 100mm, with fader eight doubling as a master fader, via a button press. There's also a reduction of Function buttons to only 12 slots, albeit providing 60 across the five layers. The display/VU screens that accompany them are a cost option; you can use the buttons without them, but it's probably not the most pleasurable experience, as the screen is the conduit to presented information. You can also build up the expanders to a 32-channel setup.

If you're really struggling for space, the P1-Nano goes even smaller. It's a one-fader device, allowing operation on a channel-by-channel basis. It's also supposedly bus powered over USB but in my tests, the unit was a little fussy about connectivity without a power supply.

» are populated with functions within the supplied template, but you can add or move buttons as you desire from within the iMap software. Even so, I think we'd all struggle to program 120 DAW-based operations that we'd need with such regularity.

As Icon have pre-loaded the most obvious commands, it's comparatively easy to do the basics, at either global or instrument channel level. How far you might choose to take this is debatable; would you access a level to then press another button to create a new project? Possibly not, if you're acquainted with the main DAW key commands.

One point to clarify is that while the Function Layers may look like a Stream Deck, they do not allow access to your computer's OS, so you won't be able to set up key macros for essentials such as "Load Pro Tools". Aren't we all doing that by voice activation now anyway?

As you get more involved with the V1-M, you do begin to find that there are some operations that are just too cumbersome for a generic control surface to handle satisfactorily. You can insert a plug-in from the V1-M, by pressing the associated Function button, before scrolling through your plug-in list and selecting your choice. The information is presented on the channel strip screen, but it's very easy to miss your preference while wading through the list of your installed plug-ins. This could be easier if you have fewer plug-ins available to you, but even with basic DAW content, that's a lot to navigate through. Once the plug-in is initiated, you can then access its associated parameters, but in the case of something like a multiband parametric, navigating the EQ can also be quite tricky. The *modus operandi* here involves the use of the infinite encoders at the top of each channel, which are more usefully thought of as stereo pan pots in their most common setting. I had hoped that by selecting an EQ band, it might be possible to adjust the frequency before pressing the aforementioned pot to toggle to the next EQ band setting in line, such as cut/boost frequency or Q. This is not the case. You navigate by using the jog wheel, while also keeping a beady eye on the information which pops up on the control surface's display, and as previously mentioned, the jog wheel can be a bit speedy!

"There is no doubt about the quality of the V1-M/X combo; these are very well built and thoroughly engaging devices, which could undoubtedly aid your workflow."

Thinking logically about this, it stands to reason that viewing information, such as a list of your installed plug-ins, is going to be far easier on a large screen. We've spent years trying to move away from the keyhole-surgery mentality, and it may be that this is simply a step too far for a humble universal control surface.

The Yin & The Yang

There is no doubt about the quality of the V1-M/X combo; these are very well built and thoroughly engaging devices, which could undoubtedly aid your workflow. There is a clear yearning from DAW users to have a greater degree of hardware-specific control, either for ergonomic reasons or because we miss the mixer concept. The V1 series goes a very good chunk of the way to fulfilling those criteria, offering something which looks, feels and acts much like a mixer. Creating automation in this environment is a breeze, as is the whole process of balancing and auditioning tracks. Building up tracks within a project is also speedy, allowing fast and simple recording and shuttling around. It's only when you delve into your plug-ins that the process begins to get a little frustrating, as navigating the plug-in hierarchy feels like hard work using such a small window on the world of a DAW project. But the things that the V1-M does well it does very well, and that's a sizeable tick for a lost mixer generation and those who want to quicken their DAW experience.

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A typical mid-gig view of Gig Performer. The main stompbox-like controls are built from a range of knobs, buttons and other objects provided in the app, all mapped to plug-ins running in the background. In this Setlist view the preconfigured list of songs and song parts can step through variant or even wildly different setups, instantaneously and glitch-free.

Gig Performer 4

Live Performance Software

Gig Performer is a live plug-in platform that's as configurable as it is crash-proof.

ROBIN BIGWOOD

Modern, powerful laptops running software instruments and effects hold lots of promise for live use. You'll need the right software for the job, though, if you don't want the experience to get messy, and that's where Gig Performer comes in. Since its first release in 2016 it has gained a reputation for both flexibility and crash-resistant robustness, and it's one of the very few applications in this relatively niche software area that's available for both macOS and Windows. What else does it offer, and could it really let you abandon your hardware on stage?

Basics

At its core, Gig Performer is a plug-in host: it'll open 64-bit VST and VST3 plug-ins in Windows, and also Audio Units on the Mac. Just as much, it's an environment in which to connect those plug-ins with the outside world, handling all sorts of signal inputs and outputs. It can use multiple channels of a connected audio interface, and multiple MIDI devices such as controller keyboards, control surfaces, pedalboards and hardware synths. It's also compatible with the OSC (Open Sound Control) protocol, letting you build custom control surfaces for it in suitably equipped third-party iOS and Android apps.

Signal flow in the program is displayed and configured with a virtual wiring view. End-to-end connections — from audio and MIDI inputs, through instruments and effects, to eventual outputs — are clearly and intuitively displayed, with colour-coded blocks and virtual, draggable wires connecting them together. The blocks have little dot symbols representing inputs and output ports, and there's nice flexibility here, with the outputs capable of splitting signals and inputs merging them, when multiple wires are connected. For more complex and ambitious setups, some dedicated utility plug-ins are provided: MIDI processors, mixers, gain controls, media players and more. Double-clicking a block opens

Gig Performer 4
$199

PROS
- Hugely configurable for individual needs.
- Robust and CPU-efficient.
- Easy to use, without being dumbed-down.
- Excellent documentation.

CONS
- Some aspects of operation feel labour-intensive.

SUMMARY

A top-class macOS/Windows application for gigging musicians that will host your software instruments, harmonise your hardware, run effects plug-ins, and much more.

controls for it in a floating window, ranging in complexity from one or two simple faders to full plug-in interfaces.

The Wiring view, however, is conceptually only the place where you build your live rigs. For on-stage use it's intended you'll work in the Panels and/or Setlists view. These typically show a simplified front end for the underlying setup, using the familiar visual paradigm of a virtual rack unit. You get to choose the appearance, building control surfaces from a range of virtual knobs, sliders, buttons, switches, labels, LEDs and meters, which are linked to plug-in parameters, or send other commands. If that sounds quite labour-intensive, well, yes, it has the potential to be. In practice it's not, though, because the trick is to expose just those few parameters you'll really need in the heat of the moment. The resulting chunky, high-vis 'large print' look is potentially then a real advantage on stage.

Panel controls (known as Widgets) can be manipulated with mouse clicks and drags, but for stage use can be driven by MIDI and OSC messages too, allowing you to tie them to a MIDI keyboard or floor unit in remote-control fashion. The relationship between a knob, say, and the plug-in parameter it controls can be complex: it could have its value range reversed, constrained, or scaled in a logarithmic, exponential, stair-step or other user-defined relationship. There are also facilities for handling hardware controllers with both standard knobs/pots and endless encoders using several different value-increment schemes.

We're nearly there with Gig Performer's core concepts, but the last few are particularly interesting for performing musicians. First, any virtual rack design, and the concoction of plug-ins behind it, is called a Rackspace. Many of these, dozens if necessary, can be loaded at one time into a single Gig Performer '.GIG' file, and while only one Rackspace is active at a time, its neighbours stay in a state of readiness. That means Gig Performer can do what various stage keyboard manufacturers call 'seamless' or smooth sound transitions — here it's known as Patch Persist — so that currently sounding virtual instrument notes (or indeed delay or reverb effect tails) are not cut off when you switch to another Rackspace, which itself will load without a delay or any audible glitches. This gives the possibility of associating different (and perhaps wildly different) Rackspaces with different songs in a set, and moving instantly between them.

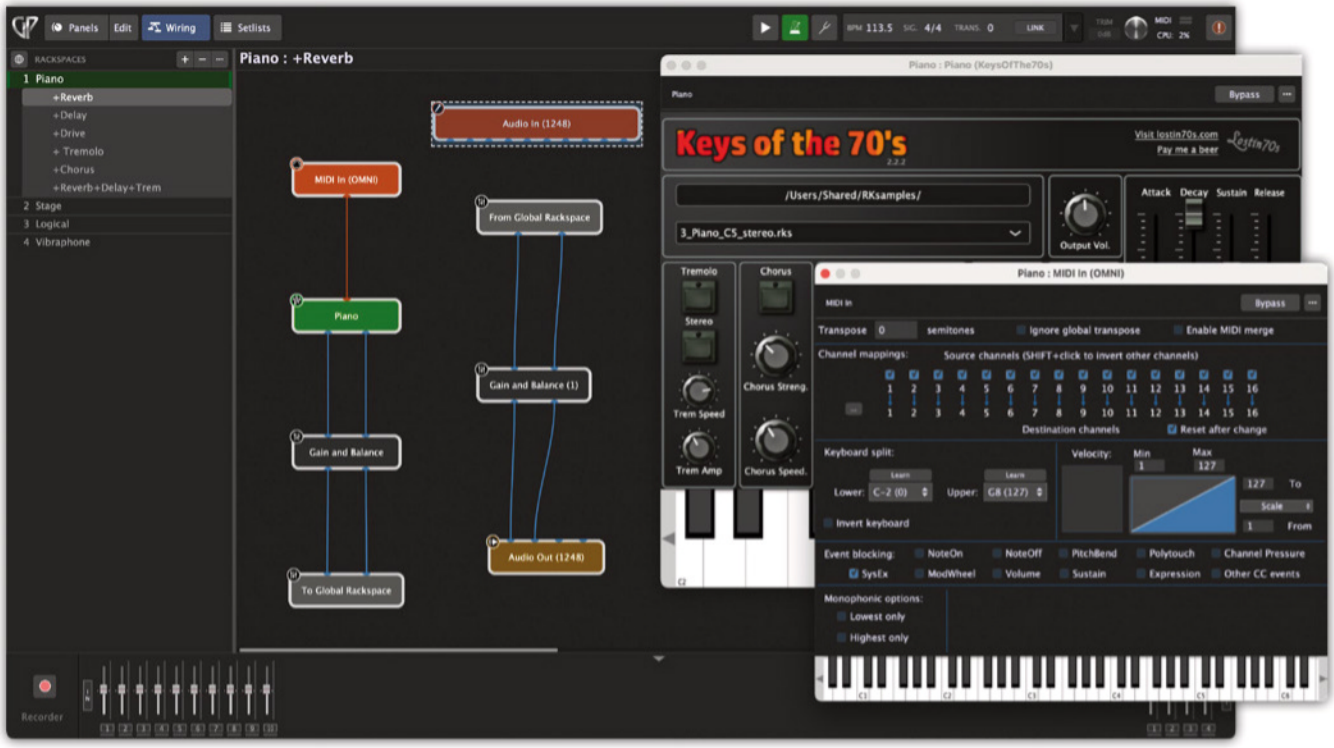
An associated feature, Variations, lets you save different settings for a single Rackspace, like automation snapshots. This comes into its own when you have a single Rackspace that does and has all you need (think guitar pedalboard for example) and you only need to bypass some plug-ins or tweak settings for different parts of a song.

Following logically on from these, there's Setlists. In another dedicated view mode, you're able to formally build a sequential list of the musical numbers (or 'songs' in Gig Performer parlance) you're intending to »

Global Awareness

Gig Performer isn't a sequencer, but it has a transport of sorts, with tempo and time signature settings and a metronome, and some always-visible (and of course, MIDI-mappable) Play and Stop buttons. These control and coordinate any tempo-driven plug-ins such as drum machines, arpeggiated synths and tempo-sync'ed effects you have loaded up. At the same time, the application can sync to external MIDI clock (though it currently can't generate it, without a suitable utility plug-in), and is compatible with Ableton Link too.

Then there's the Global Rackspace. An optional feature, this separate, master-level Rackspace has its own wiring and panel view, and is an ideal place to configure things you'll use again and again, like audio interface channel hookups, core instrument sounds, and even entire signal chains for multiple band members. Audio can then pass to and from the global Rackspace, wormhole-like, from conventional Rackspaces, using dedicated routing blocks.



What lies beneath… The same rackspace as that shown in the fist screen, in its Wiring view, with windows open for one of the bundled virtual instruments and the powerful MIDI In block, which can perform all sorts of processing and filtering, and set up keyboard splits and velocity switches.

The editing view for a Rackspace. Notice the knob selected at the lower left of the rack, the properties and plug-in mappings beneath, and the alternative 'widgets' in the list on the left.

perform in a live set, and break them down into sections (Intro, Verse, Chorus, and so on) if you like. Each of those can then be easily associated with its own Rackspace or variation, with their names unambiguously displayed in the application window along with a prompt of what's coming next. Gig Performer can also broadcast MIDI bank and patch changes to external keyboards and other devices, as songs load.

On The Road

So that's the core of Gig Performer: it's certainly extensively equipped for its role. What's it like to actually use? Broad-brush stuff first: the operational style is notably open-ended and extremely configurable. At the same time, there's good user-friendliness and first-class documentation, and I found getting going really easy, with no arcane concepts or clunky interface surprises. There's often the feel of more 'traditional' software design, with reliance on multiple additional floating windows, and somewhat varying styling across the different operational modes. That's not the modern, minimal way for touch-driven interfaces like iOS and Android, and so it doesn't surprise me that currently Gig Performer isn't available for those operating systems.

Still, for the task at hand it's what the app can do that's important, not how cool it looks, and all the basic stuff — from adding plug-ins to creating Rackspace widgets and assigning hardware controls — is quick, straightforward and clear in practice. I appreciated the fact that it's entirely possible to start with a completely clean slate. It's also great that Gig Performer addresses MIDI and audio hardware directly, and only using those channels and devices you actually need, cutting out OS-level involvement (such as macOS's Audio MIDI Setup application).

And at the risk of largely concluding this review some way before the end, I can confirm that reputation for reliability. I spent several weeks with Gig Performer trying all sorts of setups with as many permutations of internally hosted software instruments, hardware synths, audio interfaces and MIDI controllers as I could muster. I could not catch it out: it never crashed or hung, did anything weird, or failed to do something when it should. I'd have needed a lot longer to explore even a majority of conceivable setups and permutations of plug-ins, but I get the strong impression the outcome would be exactly the same. As an aside, It's notable and reassuring that there are large and active online communities of Gig Performer users reporting the same experience. User support from the company seems to be quick and helpful too. There looks to be a lot of happy people in the Gig Performer user base...

So, the basic functionality is very much fit for purpose. In addition, though, there's a lot of unseen (and somewhat unsexy) features that will be worth their weight for many users. I'll rattle through a few.

One is called Rig Manager. This adds a layer of abstraction between controls on physical MIDI gear and the software knobs and sliders within Gig Performer. Practically, it lets you do easy control remaps if you plan to (or indeed are forced to) change your hardware MIDI controller at some point, or even regularly, as you move between a home studio and live touring setup.

Then there's the way GP handles missing plug-ins and mismatching audio interface channels and models, when that situation arises. In both cases it substitutes placeholders, so you can continue to work, and save your work, without losing original settings. Whilst we're on the subject, document saving itself is unusually sophisticated. You can choose to save only individual Rackspaces, while discarding others that were perhaps unsuccessful experiments.

Even further down at the deep end of the feature set are two methods of sophisticated customisation. One is a built-in scripting language, GPScript, which can add a huge amount of functionality over and above built-in features. Some provided examples show it creating chords from single notes, building crossfaders, and it can extend MIDI capabilities too. Yet more is possible via Extensions — programmer-level stuff, but open, accessible and documented — which can rejig the user interface or build brand new functionality.

Getting back to more hands-on use, one of the most powerful features introduced in version 4 is the Streaming Audio File Player. This Wiring-view block loads up one or multiple stereo audio files in MP3, WAV, AIFF, FLAC, OGG and many other formats: useful in itself for playing backing tracks, transitional material between songs, or sound effects. However, sections can be looped, and markers added that will automatically trigger what are called Timeline Actions. They include loading

Posterity

One real coup for Gig Performer, compared to the competition, is its ability to record all the audio and MIDI of your live set as the same time as actually running it. For audio, any combination of hardware inputs and virtual outputs can be captured as WAVs at the prevailing sample rate, and in resolutions from 8 to 32 bits. As for MIDI, you get a type 1 file with a track for each physical MIDI port, and with any tempo changes embedded in the file.

Freebies

One thing I noticed after installing Gig Performer were some new plug-ins on my Mac: a VST of the guitar amp and effects simulator TH-U (or, a 'lite' version of it at least) by Overloud, and a handful of VST3s by a developer called Lostin70s Audio. Between them, they provide various guitar-leaning tone and effects options, plus some instruments. The latter include a tonewheel organ and a sample-based all-rounder with electric pianos, a Yamaha grand, Clavinet, vibes, a string machine and drums.

They're not bad at all (apart from an out-of-tune Clavinet) and several are better than things I've shelled out for. If there's a lingering sense of weirdness it's for two reasons. First, TH-U is apparently unlocked only in Gig Performer; in DAWs I had on hand, it opens with a demo version nag screen. Second, there's next to no mention of these plug-ins on the Gig Performer website or in its documentation, despite them being used extensively in the default start-up menu of demo templates.

Rackspaces, variations or songs, sending various MIDI or OSC messages, and triggering lyric or chord symbol display in a dedicated window. It's essentially time-based automation, easy to use, and with the potential to smoothly automate an entire backing track-based set. Ramp-type parameter value changes aren't supported — it's about discrete events rather than DAW-style state changes — but it has potential to do heavy lifting for quite sophisticated shows.

Surprising too, perhaps, are some sampling-related features. Two wiring blocks, Auto Sampler Generator and Auto Sampler Recorder, automate the MIDI triggering, audio recording, and subsequent file naming and saving required to create a set of samples from an internally hosted software instrument (and any associated effect chain) or indeed an external synth hooked up to your audio interface. The idea is to replace CPU-intensive plug-in chains or hardware synths you don't want to take on stage with lean-running sample replay. It can be quite sophisticated, too, with velocity switching, though no automatic looping.

Side Gig

I couldn't finish this review without mentioning a really unusual feature. From a simple File menu command, it's possible to run additional instances of Gig Performer on a single computer. Each can access its own MIDI devices, its own audio interface (if necessary, and not necessarily at the sample rate of other instances), and load its own .GIG file and Rackspaces. It gives the potential for every band member in a group, for example, to have their own personalised Gig Performer experience, as if on a separate computer, but sharing just one. Or, you might choose to run one instance for vocal effects that don't often change between songs, and another for software instruments that do. Clever stuff! Just make sure you have a computer that's up to the job.

The resulting sample file sets are generic enough to be loaded into most sample replay software worth its salt, but there's an option to automatically create a preset for the third-party freeware sample player, Decent Sampler.

Giggity

Designing software that has immense potential and flexibility but is easy to use from the get-go is no mean feat. But that's exactly what has been achieved with Gig Performer. It's blindingly obvious that it has been conceived by people who do actually play live.

Criticisms? Nothing specific or serious from me, but perhaps a few thoughts that potential users and buyers might want to take on board while they're waiting for the 14 day (almost) fully functional trial to download.

The first is about the fundamental nature of gigging software like this. When I began setting up my first soft-synth-based Rackspaces, I wondered about how presets and patchlists would be handled. In short, presets can't be changed directly from panel controls. It's neither a fault nor a weakness, and the logic behind it is sound: switching patches within a plug-in can take time, especially if sample loading is involved, and cause glitches. The solution is to use multiple Rackspaces, each with an instance of your instrument, preconfigured with the patches you need. The point is, this might require quite a shift in thinking compared to most DAW, hardware synth and effects pedal workflows, and whilst the Wiring view gives you unfettered access to plug-in windows with their preset choosers and other facilities fully intact, you'll get best results by converting to this rather more deterministic way of working, for your big night.

Similarly, there is nothing in the application that looks much like an analogue mixing desk. That's in stark contrast to one of Gig Performer's main competitors, Apple's Mainstage. There, Logic lookalike channel strips are available by default, equipped with plug-in slots, pans, faders and sends. Equivalent signal flows can certainly be constructed in Gig Performer — more complex ones in fact — and the mixer-less approach seems to be very much a conscious (perhaps guitarist-leaning) design choice, prioritising the clarity of linear signal flows over typical mixer aux/bus structures.

The flip side to that clarity, though, is that Gig Performer's Wiring view often leaves you building basic infrastructure from scratch. For example, setting up processing for an incoming vocal signal might involve adding EQ, gate and compressor plug-ins and wiring them appropriately. Easy enough, but it's unfortunate that there are not even basic versions of those amongst the bundled internal plug-ins. If you don't happen to own one in the form of a third-party plug-in (which is far from inconceivable — I discovered I had loads of dynamics plug-ins but no simple third-party EQs) then it'll be off to the Internet for you... Compare that to a Mainstage channel strip, which provides all those basic facilities by default, as well as familiar level and pan controls, and Gig Performer can definitely end up feeling harder work.

It's not a question of better or worse, and it's far from an insurmountable problem. Also it would be remiss of me not to point out that another major player in this field, the Windows-only Cantabile Performer, is in practice more or less text-based, with neither wiring views nor channel strips. So it's a matter of style and implementation more than anything.

Caveats around signal flow paradigms aside, though, I've got nothing but praise for Gig Performer. The more I used it, the more I enjoyed and admired it. Unarguably it has more advanced features than its competitors, and having spent a lot of time with it now, I certainly would trust it in a live situation. Which, speaking as a largely hardware-reliant dinosaur who's had his fingers burnt by software on stage before, is quite something.

Gig Performer is unusual, but it's exceptionally good at what it does. With laptops as powerful and relatively affordable as they are these days, it could be as good a reason as any to commit to a software-based live rig.

$ $199
W www.gigperformer.com

Donner Essential B1

Analogue Synthesizer & Sequencer

Fatboy Slim's dream of universal 303 ownership inches a little closer with this cut-price acid box.

ROBIN VINCENT

The Donner B1 cut a handsomely futuristic form on my desk. The black and titanium finish oozed class, and the blue and white lighting gave it a cold and serious vibe that felt out of place once you set the synth in motion. When I pressed Play, the synth came alive with a repeating pattern of familiar and pleasing bass lines. I sought out the filter controls and found that I could quickly push the sawtooth and square waves into a wonderfully juicy squelch. A simple decay envelope pulled at the filter, giving it a penetrating zap as I drove into the resonance. There was definitely a tart, acidic tone going on here. Wait a minute... was this a 303?

Yes, the Essential B1 Analogue Syn Bass is Donner's take on the classic Roland acid synth, and we can never have enough of those. And honestly, I don't think I'm done with it yet. Something about the bounce, the rip, the squeak and squelch is so beguiling that I couldn't help but have fun with it. As to the question of whether it sounds like a 303, all I can say is that I've had it sat next to the Behringer TD-3, and other than the B1 being a lot louder, you can't tell them apart. So, from one analogue clone to another, it sounds as good as it should.

The influence of the 303 will no doubt linger, but I wanted to try to review the B1 on its own merits. Donner have helped with this by ditching the 303 sequencer entirely, giving me something new to critique. And they've added some effects and interesting performance features. A recent firmware update has also introduced a Song mode, addressing a lot of the criticisms that have been levelled at it since its original release in May 2022.

Specs

The analogue synth engine and signal path have a single VCO with two waveforms,

Donner Essential B1
$169

PROS
- Sounds like a clone of a 303.
- Built-in saturation and delay effects.
- Easy-to-use sequencer.
- Fun performance features.
- Song mode.
- Classy styling.

CONS
- All the important knobs are a bit cramped together.
- Sequencer is not as refined as it could be.
- Sequencer controls would benefit from a better layout.
- Not powered by USB.
- Over-exuberant random generator.

SUMMARY

The Donner B1 offers a 303 synth-type experience with none of the sequencing headaches. It has nice performance features, effects and stylish looks, which bring some functional challenges of their own. Imperfect, but great fun.

a low-pass VCF with cutoff and resonance controls, and a VCA with a simple Decay envelope. The envelope runs the filter and the VCA if you hold down a note. An Accent control boosts a step if it's activated in the sequencer. The Pitch knob bends the oscillator up or down a fifth. That's your top row of nondescript knobs.

Along the second row are an analogue Saturation effect with Drive and Tone, and an analogue Delay with Level, Time and Feedback. Below are a one-note-more-than-two-octave button keyboard and a sequencer control panel with a three-digit display.

To the right is a larger volume knob. To the left are four patch sockets: Aux in for mixing in an external source (this doesn't go through the filter), Headphone output and Sync In/Out. On the back, we find a 5-pin MIDI DIN In and Out, a mono quarter-inch jack output, a power socket for the included supply and a USB-C port. The B1 has to be powered by the adaptor; it won't power over USB and has no battery compartment.

Form

Donner have gone for a very clean and ordered design. It looks smart, although the metallic illusion falls away when you pick it up. It's light and plastic, but it feels solid enough, and the raked angle looks really nice on the desk.

All the important synthesizer bits only take up the top third of the front panel. The knobs are small but adequate, although they're a little cramped for my fingers. I've seen some people swap the Master and Cutoff knobs to give the one control you will be using all the time a bit more ballast. But, in my view, that makes things more cramped.

Most of the space is given over to the 26-note, strangely oblong keyboard. The extra note lets it pull off a full 16 steps on the 'white' keys for the purpose of sequencing. The keys are a bit like sliced-up drum pads. There is no velocity at play, and they light up white when you press them, which takes more effort than you'd think. I initially thought that too much emphasis was placed on the keyboard, giving it more space than needed at the expense of the cramped synth controls. If this was a 303, I felt, you'd spend all your time on the knobs, not the keyboard. But the keyboard brings a lot to the table, pushing the B1 away from its 303 roots. It encourages you to play it as a synth, to switch on the delay and enjoy the length of the envelope like you might on other synths. There's also an arpeggiator that handles as many notes as you press. So the keyboard is definitely taking on a larger role in the B1, but I still can't help feeling that Donner might have given the synth controls more room.

Sequencer

I'm not too proud to admit that when I sat the TD-3 next to the B1, I spent 10 minutes fiddling in futility before I googled "how to write a pattern on the TD-3". The B1 is a dream in comparison. I pressed Rec/Edit, played the notes, and I was done. If I wanted to put in a rest, I pressed the 'Rest' button. If I wanted to add a Slide or Accent, I did that while holding the note. It was 16 steps of easy step sequencing. While I believe classic acid bass lines can only be forged during the pain you experience using an authentic 303 sequencer, I'll happily take this easier, if less heroic, path.

Once you have your sequence, there are a few things you can add to each step. In Edit mode, the steps all light up in an inviting blue colour, but don't be tempted to touch them to select a step because all you'll do is change the note for the selected step. You have to use the up/down arrow buttons to navigate the steps. The steps flash as you move through them, and the note is shown by the top half of the key lighting up white. Displaying the notes is one of the firmware improvements on the original release, and it's very helpful indeed.

In addition to Slide and Accent, each step has a Gate Length and Ratchet value. Gate Length goes from zero, which is a rest, up to eight, which ties it to the following note. You can have up to four Ratchets. With the step selected, you can hold the button and use the arrow buttons to set the value. When you go back through the steps, there's no indication of a Ratchet or changed Gate Length being applied, which I found slightly odd because the Slide and Accent buttons illuminate when they are active on a step.

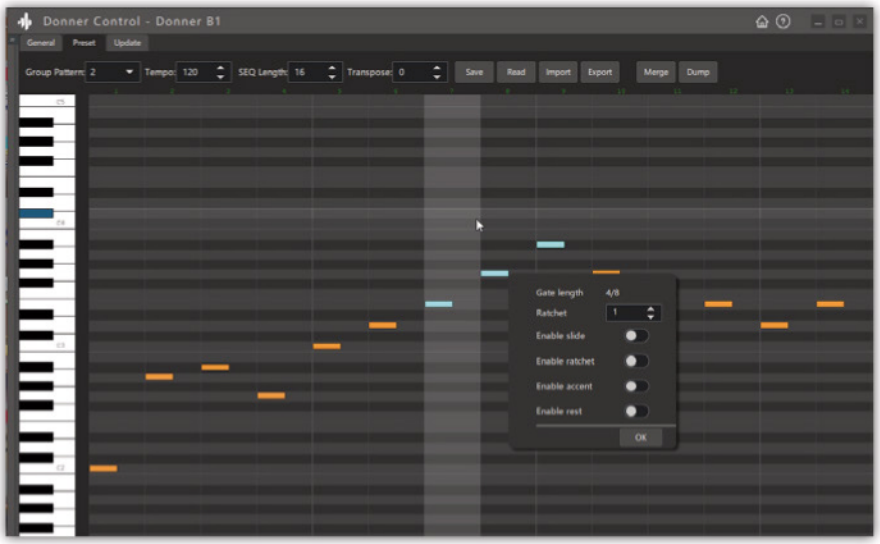
Once I was over the initial relief that the sequencer is easy to use, I found it is far from perfect. It's not possible to select a step simply by pressing on it, which is frustrating as they are right there in front of you. In Play mode, you can hold Rec, select a note, and then add Accents and Slides — so why can't you do this in Edit mode? You can also turn steps on and off in Play mode, but not in Edit mode. However, you have to be careful, because pressing a note in Play mode transposes the sequence. To turn a step off, you have to press and hold it until the light goes out. Transposition is a great feature, but it's too easy to do it accidentally while you're trying to turn off a step.

The numeric-style pad that controls the sequencer is not awesome. I too often did the wrong thing and hit the wrong button. It's impossible to read in low light, and those arrow keys got right up my nose. The transport controls would benefit from being elsewhere and perhaps from being upgraded to big green and red buttons. I think the usability has suffered somewhat because the design department wanted the B1 to look really ordered and precise.

There is a Donner Control app, which runs on a computer and gives access to MIDI settings and other parameters. It also has a piano-roll editor for the sequencer. It's not a real-time thing; you have to suck the sequence out of the B1 and then blow it back once you've edited it. You can also use it to store presets and import them from elsewhere. I had some trouble getting the B1 and the app to see each other, but once I'd done the right combination of switching things off and on, it functioned fine.

Performance & Song Modes

All of the step parameters are available as performance features that you could drop »



The Donner Control app provides a non-real-time piano-roll sequencer and takes care of MIDI setup and patch storage duties.

» in on the fly. These are pretty great and give you a whole extra sense of interaction. As the pattern plays, you can drop in Ratchets, Slides, Accents and enforce Gate Lengths. You can also use Clear to mute the playback and Hold to retrigger the current step. Without a doubt, it's a lot of fun. That is, until you accidentally hit the Arp button right there in the middle of them, and it instantly stops playback. That control pad definitely needs a rethink.

The significant new feature in the 1.10 firmware update is Song mode. The basic idea is that you have 16 slots into which you can place any of the 128 patterns. The 16 slots then play back in order, giving you your song. It works and means you can play back something more complex, or simply treat it as a way of extending the step count.

You have to stay in Song mode for it to follow the plan. So you can't 'see' the patterns play as you would in Play mode. And if you drop out of Song mode back into Pattern mode, you can't get back without stopping playback. You can disengage Song mode from within Song mode, so it spins on the current pattern until you engage it again, which is pretty useful when performing.

All the performance features still work in Song mode except for the Clear button, which no longer mutes the playback for some reason.

Random Notes

Another new feature is the ability to generate random notes. It's quite clever in that if it's an empty pattern, it will generate a random sequence, but if it's not, then it will keep the sequence and only randomise the Accent, Slides, Gates and Ratchets. In either case, the performance features get splurged across the sequence in a wodge of acidic messiness, and that's a real shame. If only there was a way of generating notes without all the bells and whistles, because it currently feels too epic for me and ultimately unusable.

In Flight

Despite all my niggles about the sequencer, the enjoyment level when running a sequence is off the charts. The playfulness of the 303 sound engine is legendary, and you absolutely have that here.

Switch in the saturation, and it screams at you in the most delicious way. The delay gives you everything from bathroom reflections to about half a second of repeats. The maximum level is only about 50 percent mix, and maximum feedback gives you about 12 echoes. It's not spectacular; it sounds a bit dull as it shaves off all the top end, but it's fun and valuable to have.

However, it's the performance features that really give it a life beyond the 303. They give you a lot more to do and, in combination with the Song mode, offer a great deal of variation and versatility. So, unless you crave the authenticity of a 303 sequencer, then the Donner B1 is a great choice for a budget, analogue acid synth. ■■■

$ $169
W www.donnermusic.com

The B1's layout places aux in, headphone and sync I/O ports on the front panel, while USB C, MIDI I/O and a quarter-inch audio output are found at the rear.

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Soothe2 is a dynamic resonance suppressor. It identifies problematic resonances on the fly and applies matching reduction automatically. This results in a smoother, more balanced sound and saves you from having to notch out frequencies by hand.

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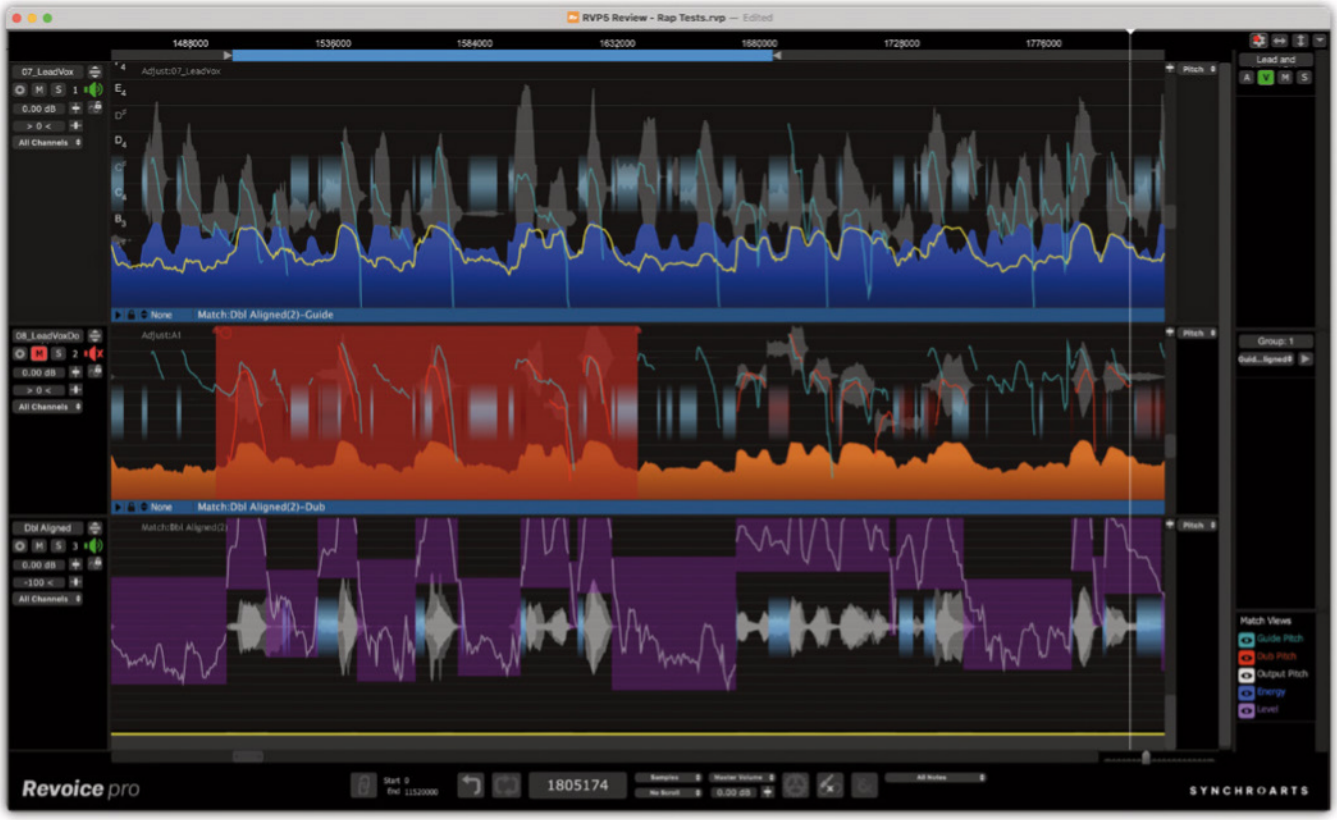
Synchro Arts Revoice Pro 5

Vocal Alignment & Pitch Correction Software



MATT HOUGHTON

As this vocal production powerhouse evolves, it's not only growing more powerful — it's also becoming quicker and easier to use.

At heart, Revoice Pro (I'll refer to it as RVP from here) is an audio pitch-, formant- and time-processing app. There are plenty of those available now, including some powerful options built into DAWs, but what makes RVP particularly interesting is its unique ability to analyse multiple audio files and then apply one or more characteristics (pitch, timing and/or level) of one of them to any or all of the others. For instance, if you have two singers singing in unison and you wish to tighten up the pitch and timing differences between them, this can be achieved in just a couple of clicks. It doesn't take much more effort to do the same for a whole stack of backing vocals, whether they're singing in unison or harmonies. And, applying the same concept 'in reverse', RVP can create some very natural-sounding fake double-tracks too, by cloning a vocal part and then applying a specified amount of randomness to the clone's timing, pitch and level. Furthermore, the results of any process can be tweaked for every new part created, and the results of all these processes can be dynamically linked, so that if you go into the Guide track and make corrections to the pitch or timing, these will all be cascaded out to any Dubs or Doubles you've created.

It's pretty powerful stuff and, compared with traditional processors and editing tools, it has the potential to save music producers a huge amount of time when working with vocals, in almost any genre. And despite the name, it can also be used with other monophonic sources. I've used it to good effect with lead guitars and bases, for instance. And it's not just about speed and convenience: importantly, the processing also sounds very good.

Almost unbelievably, it's approaching half a decade since I first evaluated RVP4, and if you've already read my review (which is free to access on the *SOS* website:

Synchro Arts
Revoice Pro 5
$599

PROS
- Still unique.
- Sounds as good as ever.
- Slicker pitch/time manipulation tools.
- Smoother integration with your DAW.

CONS
- None.

SUMMARY
For vocal pitch and time processing, Revoice Pro remains in a class of its own — especially when it comes to creating doubles and working with backing vocal stacks.

https://sosm.ag/synchro_arts_revoice_pro4), you'll know that I ranked it amongst the most natural-sounding pitch- and time-manipulation processors available at that time. It still is now.

Nonetheless, there were some aspects of v4 that I felt might be improved. Despite progress since earlier versions, moving audio between some DAWs and RVP could feel a little clunky, and inside RVP I found some of the terminology a bit alien, while some processes took more clicks and keystrokes than I felt should be necessary. Well, step forward Revoice Pro 5 — this latest version does a lot to address all those issues, while also ushering in an abundance of helpful changes that should make the app quicker and easier to use for everyone.

Plugged In Thinking

RVP5 is, at heart, a standalone application (for Mac and Windows OS) that can be used completely independently of your DAW and, as with previous versions, is authorised via iLok. But installed along with the main app are a few DAW plug-ins that aim to take the pain out of transferring audio from your DAW to RVP and back again. It's perhaps worth noting that the old AudioSuite plug-ins for Pro Tools — the ones that allow you to apply presets pretty much instantly from within the Edit window — are still present, and that RVP still supports the drag-and-drop transfer of audio clips between applications. But the big news is that the Revoice Pro Link plug-in, available in AudioSuite, ARA2 and AU/VST3 flavours, has been reworked to make it quicker and easier both to get audio from your DAW into RVP, and to monitor the result in your DAW.

More DAWs support ARA2 now than when RVP4 came out, and I expect this version of the plug-in will be what the majority of users end up using most. Depending on how your DAW handles ARA2 and what you're trying to do, you might prefer to instantiate this plug-in at the DAW track level, where it can be used to capture/process/monitor multiple different clips, or directly on individual audio clips. Once instantiated, the GUI will tell you if RVP isn't open (you're prompted to click a button to open it). You can then hit a button to capture the selected clip(s) into RVP and spot it on the RVP timeline, so it's in sync with your DAW — you have the choice to specify to which RVP track it's captured, or to automatically create a track. Now, when you press play in your DAW and the playhead gets to that clip, you'll be monitoring the audio for that clip in RVP, rather than the original that still sits on your DAW's audio track. When ready to commit, you just use your DAW's render function, as you might with any other effect or virtual instrument.

Something that's new is that you can now, from the ARA2 plug-in GUI, do more than simply capture the audio into RVP. You also have the option to apply some presets: instead of Capture Only, which is the default, you can choose Capture and Match Timing / Pitch / Level or Capture & Create Double(s).

There's a very similar AudioSuite version of this Revoice Pro Link plug-in for Pro Tools users, and if using ARA2 is inconvenient for you, or your DAW doesn't support it, you still have access to an AU/VST3 version. This is similar, but records the audio during DAW playback and doesn't offer the 'instant processing' options, only the capture and monitoring. While playing the audio in does mean a slightly longer wait, performing those processes inside RVP isn't daunting.

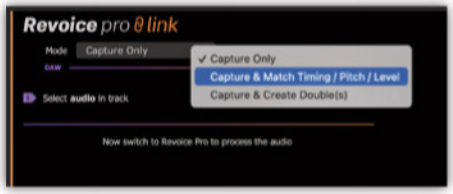
There will also be times when you are performing more sophisticated processes in RVP that create new audio files, and these will often require their own DAW tracks. Likewise, the way your DAW handles mono and stereo tracks can make it tricker to monitor what you want in the DAW. In either case, you can simply select the desired clips in RVP, hit Shift+Option (for Mac — I believe it's Shift+Ctrl for Windows), drag them into your DAW, release the modifier keys and unclick. Then just use the relevant command in your DAW to 'spot' these time-stamped clips on the timeline (in Cubase, for example, this is called Move To Origin).

It's worth pointing out that Synchro Arts have published an excellent series of YouTube tutorial videos to help users of different DAWs get their bearings. And what this all adds up to is that RVP5 feels much better integrated into your DAW than previous versions: you'll spend less time setting up RVP, and less time switching between the two applications. I encountered one or two very small bugs with the transfer and monitoring that had squeezed past beta testing, but nothing that prevented me using RVP comfortably, and Synchro Arts assure me these are being dealt with.

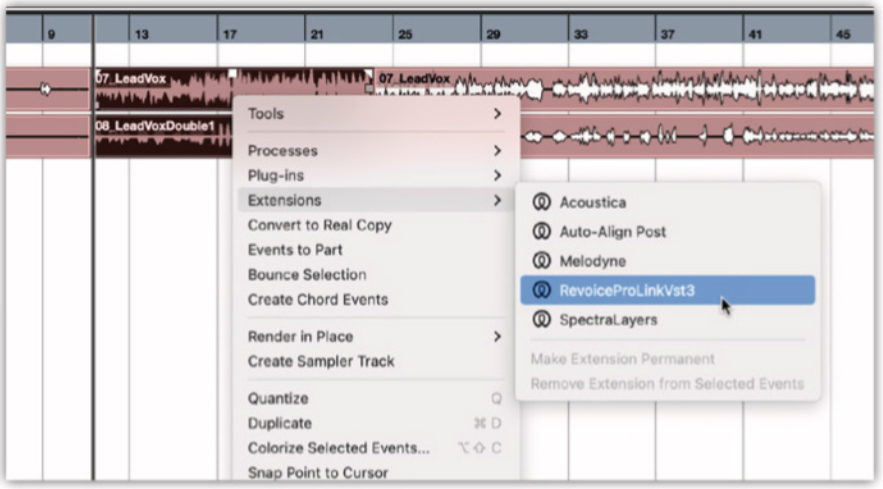
Power Tools

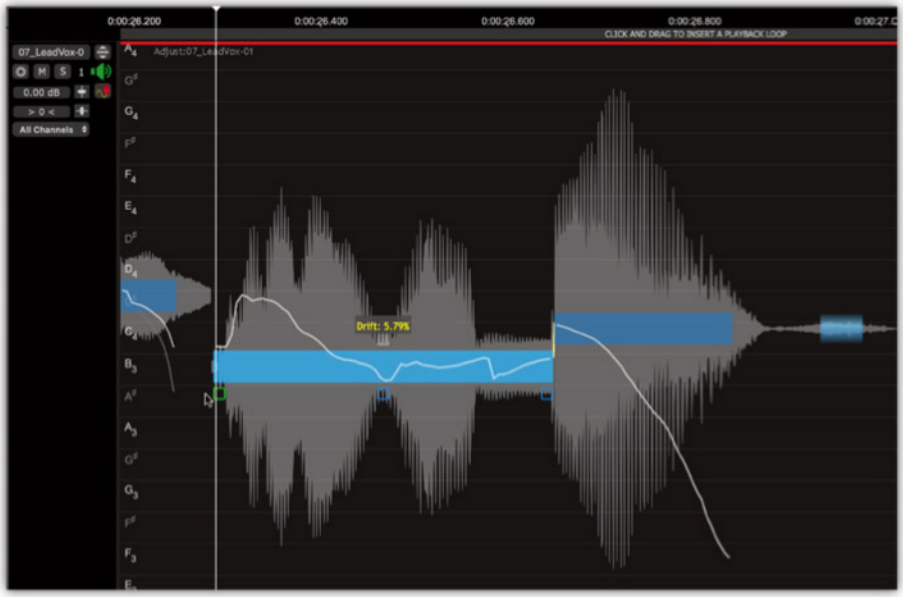
RVP4 had a number of useful tools for editing the pitch and time of a part, and these all remain present and correct in v5. There are helpful tools for splitting note blocks, for adjusting the pitch automatically or manually, altering pitch drift, time-stretching and more besides. Of course, while RVP is great for detailed edits, it's also capable of automatic broad-brush correction. For instance, to snap a whole part to the nearest note, all you need to do is hit Cmd+W (Ctrl+W in Windows). Unlike if you're using a real-time automatic tuning plug-in, »

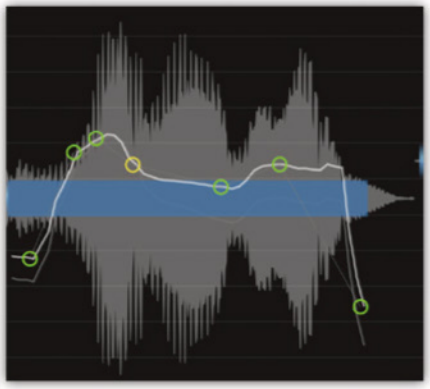
The new ARA2 plug-in allows you to do rather more than simply transfer audio from your DAW to RVP.



Instantiating the ARA2 plug-in on specific audio clips in Cubase.





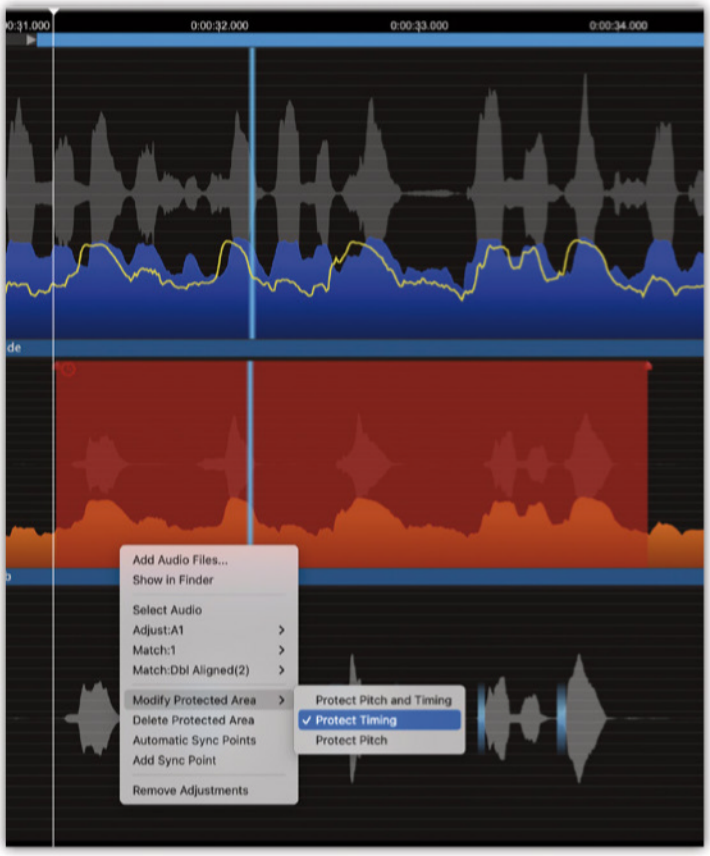


Alongside the existing tools, Revoice Pro 5 has the more versatile and intuitive tools Synchro Arts introduced in their RePitch software, with adjustable nodes on note blocks and the Shaper tool.

» you can also then go in and tweak the details to the nth degree. And when it comes to doing that, there's been a substantial improvement...

The main 'warp' view, in which you can adjust the pitch, timing and level of a whole part or individual notes, has been treated to a significant overhaul. In essence, the old Selector tool has been turned into a versatile multi-tool — a feature borrowed from Synchro Arts' RePitch (released about a year before RVP5, and reviewed by Sam Inglis in *SOS* February 2023: https://sosm.ag/synchro-arts-repitch). Detected notes are represented as 'blocks' as before, but each block now has four nodes that act as handles for different processes. The mouse cursor's purpose also changes according to where on the block you place it before clicking. By the way, RePitch's superior zooming and scrolling features for the main page have also been inherited.

The nodes allow you to click-drag to flatten, expand or invert the pitch modulation within a block's pitch (top middle node), or adjust the pitch drift (bottom left), level (bottom middle) and amount of pitch correction being applied (0-100%). Move the cursor elsewhere on the block, and you can click-drag to move or time-stretch the note, or adjust the pitch. Meanwhile, double clicking on a block forces the note to 'zero'. The Draw tool already allowed you to adjust a block's pitch trace manually, but with the new node-based Shaper tool it's super easy to smooth, exaggerate or completely rewrite a note's natural modulations and vibrato. What's more, RVP detects non-pitched elements, such as noise, breaths, esses and certain consonants, and treats them differently: the block will have only one node, which controls the level. This can be really helpful if you like to de-ess manually as you go through a part to work on pitch and timing. The changes are very welcome, as they mean RVP feels significantly quicker and easier to use.



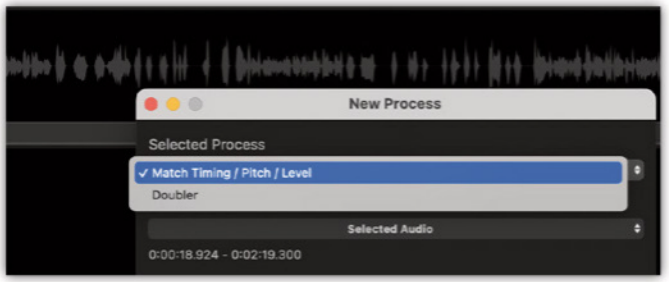
Revoice Pro is pretty good at time-alignment, even when there are gaps in the audio. But if you experience problems, you can specify regions of a Dub part that you wish to be immune to any processes being applied. These areas appear highlighted in red.

Processes & Presets

The *raison d'être* of RVP extends well beyond the processing of individual files, though. The cleverest part is the way it can apply the characteristics of one file onto others and create fake doubles. There's been strong progress here too. First, the best way to get what you want from RVP processes has always been to choose a preset and tweak it, and there's now a greater array of presets to choose from. Partly, that's because there are new features, and chief amongst those is probably SmartPitch. You can, as before, get RVP to force the output from aligning a Dub part to the same note as the Guide, either in the same or the nearest octave. And

Terms Of Engagement

Something existing RVP users will notice, and prospective ones should welcome, is that there have been a few changes in the terminology used in Revoice Pro. For example, the process of applying the characteristics of one track onto the audio of another used to be called an Audio Performance Transfer, or APT for short. Now, it's described more simply as Match Timing / Pitch / Level. It's a welcome change — I recall finding the APT term confusing when I first started using RVP, and everyone should be able to understand what it does now, whether their background is in dialogue replacement (where Synchro Arts began) or music production. Similarly, the function formerly known as Create Warp Region is now more prosaically referred to as Adjust Pitch / Timing / Level. Users of previous versions needn't worry, though: the default shortcuts, some with their roots in the old terminology, remain unchanged: you can still hit W (for 'warp') if you want to start editing the pitch, timing or level of a part in detail (and you can still customise key commands, too).



Synchro Arts have renamed some functions, making their purpose more obvious. For example, the Audio Performance Transfer is now called Match Timing / Pitch / Level.

you could tune the Dub individually if preferred. But now, RVP can work much better with harmony parts. SmartPitch decides where it should to force a harmonising double to the same note as the Guide, and where simply to correct to the nearest semitone or scale note. For me, this worked flawlessly and is a big reason to upgrade from v4.

RVP can also do a lot to help you when working with what I think of as 'intermittent doubles'. That is, vocals that come in to accentuate some phrases in the lead vocal but not others. I remember working in RVP4 with a rap track in which that technique was used a lot, and I found it more challenging than I'd hoped. When working with a single file, the long periods of silence apparently caused RVP some confusion, throwing its time-alignment out of whack. Already, in RVP4, Synchro Arts had brought in the SmartAlign feature to deal with this, but while this has worked very well on most material I tried, on a couple of specific projects the audio seemed to 'trick' RVP into sync'ing the wrong points in two files. Thankfully, there were ways around this that I probably hadn't fully got my head around when writing that review, and these facilities remain in v5. You can, for example, 'protect' certain areas of a dub from the applied processes, and you can also create sync points that instruct RVP where a certain point in the Dub should match a corresponding point in the Guide, just to make the algorithm's life a little easier. These also remain in RVP5, but I think the new Revoice Pro Link plug-in, and particularly the ARA2 version, will also make it quicker and easier to go through your project and treat such 'intermittent' parts as separate clips in RVP. I should stress that, for the most part, RVP does a cracking job of aligning multiple signals — a better job than anything else I've used.

The Bottom Line

Revoice Pro has always been unique and incredibly powerful but, in version 5, Synchro Arts have delivered a product that feels mature. There's potentially much more I could have written about the finer details when using it with different material and different DAWs, but the manual and Synchro Arts' tutorial videos can fill in most of those blanks. Meanwhile, I hope I've managed to convey a sense of just what Revoice Pro 5 could do for your productions, and how much more intuitive it is than its predecessors. It may not be the most affordable app for home studio producers, but this is Synchro Arts' flagship, do-everything product. Also, the upgrade price from v4 is actually pretty modest, and if you don't need all the functionality then the more affordable Synchro Arts apps that have a narrower focus, such as VocAlign and RePitch, might be worth checking out too. Highly recommended.

$ New license $599 or rent to own (four months) for $180 per month. Discounts apply for owners of v4 and other Synchro Arts software.
E sales@synchroarts.com
W www.synchroarts.com

Electro-Harmonix Pico Deep Freeze

Sound-sustaining Effect Pedal

Many of EHX's Pico pedals are stripped-back versions of their larger counterparts, but the Deep Freeze, which comes with a PSU, combines features of their Freeze and Superego pedals. It's designed to freeze a short section of sound to produce a smooth sustain, and it works on both single notes and chords — so instruments such as guitar or piano can be used to generate drones or pads that sustain indefinitely, with the option of playing the dry sound over the top.

To the original Freeze's feature set, the Deep Freeze adds layering, adjustable attack/decay speeds, Gliss, dedicated wet and dry volume controls, and three distinct operational modes: Latch, Moment, and Auto. Latch freezes and sustains sound when the footswitch is pressed until it's pressed again. Moment freezes things only while the footswitch is held down, while Auto freezes the sound automatically when the input signal level exceeds its internal triggering threshold. Other than the Effect and Dry volume controls, there are just two further knobs, a Mode button and the footswitch. The current mode is indicated by the LED colour.

Gliss sets the transition time between freeze sounds, and at extreme settings, can produce some endearingly weird results. The Speed/Layer knob functions differently depending on which mode is active. In Moment mode, Speed controls attack and decay times simultaneously, while in Latch mode it adjusts the amount of layering, allowing the user to build up chords. In Auto mode, it sets how long it takes the frozen sound to fade out (but can be switched to control the rate of fade-in). To switch between the Decay and Attack modes for the Speed knob, press and hold the Mode button until the LED colours cycle. Using a power-up sequence, the bypass mode can be set to be digital, analogue or hybrid, the last of these automatically switching from digital to analogue when there's a gap in the input signal.

Sonically, the pedal produces a reasonably smooth sustain that provides a useful pad-like backdrop over which you can play, though Auto mode opens up more creative possibilities by allowing you to make changes to the frozen sound just by playing new notes or chords. Auto, though, is the least natural-sounding mode: when using the other modes it's usual to time your pedal action to be just after the note attack, but as in Auto part of the note attack can sometimes be looped in the frozen sound, it can create a weird reverb-like effect; this is most noticeable at short decay settings. Weird can sometimes be be good, though: experimenting with the Gliss and Speed/Layer knob is key to fine-tuning the weirdness, and subjecting the frozen sound to effects such as modulation or delay can also be rewarding. As a stepping stone between the Freeze and the Super Ego, the Pico Deep Freeze has much to commend it. *Paul White*

$ $193.10.
W https://www.ehx.com

Mixwave Coil Audio CA-70S

Saturation Plug-in

Coil Audio's CA-70 valve preamp is based on classic Western Electric and RCA designs dating back as far as the 1930s. Unlike later preamp circuits, these used the valves as fixed-gain elements and the overall gain was adjusted using attenuation on the front or back end. As the CA-70 has two valve gain stages, it's designed with two attenuators: a stepped input pad, and a variable control that adjusts the level passing from the first stage to the second. It also has tone-shaping filters and a dial that sets the amount of negative feedback. Available in several physical formats, the CA-70 has proved very popular since its launch a few years back, and has now inspired an official plug-in emulation.

Mixwave's CA-70S faithfully replicates the rackmount version's feature set, with a choice of input sensitivity settings, a five-position Low switch that affects the tone of the bottom end, plus feedback and 'output' controls, the latter being in fact the inter-stage gain control described above. On top of this, they've added controls to help it fit into a plug-in context, including fully variable input and output trim, optional high- and low-pass filters on both input and output, and a wet/dry mix knob.

Contrary to some people's expectations, valve gear isn't intrinsically colourful or distorted. When used within its intended operating range, it can be very clean, and that's reflected in this plug-in. If you operate it in 'line' mode, leave the negative feedback control in its default position and feed it a signal with a reasonable amount of headroom, you're unlikely to hear significant changes. As you begin to play with the controls, you discover that it's certainly possible to achieve crunchy drums and overdriven telephone-style vocals, but what's perhaps most impressive is the huge hinterland of 'subtle warming' treatments that lurks between these extremes. This definitely isn't one of those saturation plug-ins that jumps straight from "Is it on?" to "Too much!"

The effect of the Low control is rarely dramatic and, as you'd expect, it's mainly useful on sources with a lot of low-frequency content. In conjunction with the NF dial, it makes the CA-70S plug-in a brilliant tool for treating synths and electric basses. If what you're looking for is rich, thick, controllable warmth that never gets harsh or brittle, this is definitely the plug-in for you. On sources with more midrange and high-frequency content, things can get a bit edgy at one end of the NF control's travel, whilst turning the feedback all the way up delivers a muted, darker tone, but pretty much the entire range has the potential to be useful somewhere. Its relative restraint makes it usable on busses and entire mixes as well as on individual sources.

Mixwave have also sourced a preset collection from an impressive roll-call of engineers and producers. Step through these and you'll hear one or two that seem much more radical than the others. They typically make use of the input or output filters, which are the only controls really

capable of extreme results; however, unless you click to expand their control panels, there's no visual indication of whether they're active, or what frequencies they're set to. This could perhaps be improved in a future version. Other than that minor foible, this is a very classy software implementation of what seems a very classy piece of hardware — and, naturally, it's an order of magnitude cheaper than the physical CA-70. *Sam Inglis*

$ $149 (discounted to $99 when going to press).
W https://mixwave.com

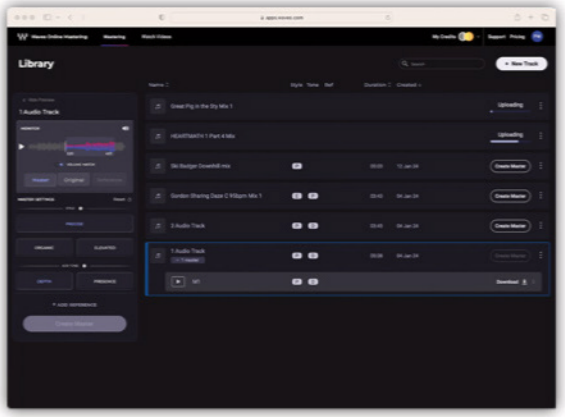
Waves Online Mastering

AI-assisted Mastering Service

Online automated mastering isn't a new idea — LANDR have offered such a service for several years — but Waves have taken a different approach from most. They've used modelling and machine learning to draw on the gear, ears and decision making of award-winning mastering engineer Piper Payne. The resulting process attempts to apply the same sort of processing and judgements to your tracks that she would, and aims to ensure that the mastered track translates well on any playback system. In addition to the usual Mac and Windows support, Waves Online Mastering will work with Android and iOS. You can try the system for free, and only need to pay once you're happy with the results.

The process is fairly straightforward: you can drag and drop your mixed track(s) into the Waves Online Mastering window, and they show up in the main body of the screen. Your audio gets uploaded to a secure server, and a short time later you can access a 30-second preview section of your track, with buttons to switch back and forth between the original and processed version. As a mastered track is usually louder than the original, there's a loudness compensation switch that lets you hear the track pre- and post-processing at similar levels. The mastering engine creates a new 30-second snippet for every mastering revision, and if a reference song is being used, then these preview snippets match to the reference.

You don't need to select a musical genre, but can opt for different mastering settings. Precise is the one that delivers optimised settings for your track based on the algorithms behind Waves Online Mastering, but you can select Organic, which dials back the processing, or Elevated, which takes the processing a little further. Additionally there are buttons for Depth and Presence, which add a little bass or treble lift when active, and both can be used together if desired. Once you've made changes in Preview mode, they will be added to the final master once you commit to it.



If there's a commercial track that has a similar style, timbre and dynamics to your own mix, you can import that as a reference. In my experience with standard match EQs (admittedly much blunter tools), it helps if the reference track is in the same key and covers the same general musical range. Doing the same here is advised, as the mastering engine references both the EQ spectrum and the overall loudness of the reference.

Once your settings are complete, clicking on Create Master starts the process and until this point there's no charge. After your first free mastered track, you have to purchase credits: one credit is needed for each track you master, but you still don't need to pay until you commit to a master. The user can view their secure library of songs at any later date, play full masters stored in the library and download masters in any supported format with a choice of sample rate and word length. It's also possible to create new masters of additional revisions.

There are some things you don't get here that other mastering tools offer. Firstly, there's no EQ curve visualisation — this isn't *needed* when using the service but can be a helpful in highlighting common EQ traits in your own mixes. After all, the closer you can get to a mix with a well-balanced spectrum, the less EQ the mastering tool has to apply. There also seems to be no way to set a target loudness in this version or to fine-tune the stereo width, and you can't adjust how much influence a reference track has. Talking to Waves, it seems that these were deliberate decisions to keep the process as simple as possible for musicians who are less technically inclined.

With my own mixes, I used Logic's Loudness meter to check the final levels and found that some hovered around the -7 LUFS mark, which is a little hotter than I usually like. One of my tests was to master some very old mixes I'd made of my band in the late 1970s, all recorded using a Tascam four-track machine and some very cheap mics. The originals were somewhat thin-sounding with a slightly abrasive edge, but after mastering they sounded gratifyingly well balanced and punchy, as well as smoother at the top end. It's probably safe to say that most modern tracks submitted for mastering will be in better shape than these, and processing some more recent mixes revealed that, other than changes in loudness, the overall treatment was far more subtle — yet it still applied a welcome polish to the sound.

I have to conclude that the end results generally are impressive, both as regards tonal balance and dynamic range adjustment. In comparison with Logic Pro's inbuilt mastering, it comes across as perhaps slightly smoother-sounding, with more assertive dynamic range control. Detail is lifted out without adding undue harshness while bass-light mixes are given additional heft without making them sound boomy. Yes, I'd personally have liked a little more control over proceedings but, for most types of music, Waves Online Mastering comes up with something that compares favourably with all but the most sophisticated professional mastering.

Invariably, any fully automatic process will work better with some mixes than others but, as my test with ancient mixes confirmed, this mastering engine is capable of making sensible processing decisions and it should prove an effective tool for the target market, which I see as mainly semi-professional music creators who want to get their mixes into decent shape to put online, or for studio professionals who want to give their clients a listening copy of their mix that will be much closer to the final mastered version than a straight render. Given that Waves will do a fine job of polishing your mix for less than the cost of a glass of beer, there's little harm in trying it and very little to complain about. *Paul White*

$ From $2.99 per track (60 credit package) to $5.99 for a single track (1 credit).
W www.waves.com

When you're tracking live in the same room, drum spill onto the vocal mic is inevitable!

MIX RESCUE

Robin Phillips

We help transport listeners from a small studio to the Big Easy!

SAM INGLIS

Robin Phillips is a very fine pianist and singer, who is well known on the London jazz scene. His home-recorded piano skills are currently earning millions of streams for jazz-house artist Berlioz, while his vocal chops lent some class to our video feature comparing versions of the AKG C414 (www.youtube.com/watch?v=B7NssHrswlU). And late last year, Robin was wrapping up a project that was particularly special to him: his first album of original songs for over 20 years. This latest endeavour features not only regular bandmates from a number of his current line-ups, but also London's Soul Sanctuary gospel choir, big-name jazz players from both sides of the Atlantic, and a string quartet. The first single from the album is 'Ode To NOLA', his homage to New Orleans.

Robin normally mixes and masters everything himself, but this particular track wasn't cooperating. Given its importance as the lead-off single from the album, he called me to ask for some advice. One thing led to another, and soon I had a very neat and well-organised multitrack to download!

Close Calls

Visits to historic studios such as Sun Studios and Muscle Shoals Sound Studios had convinced Robin that recording musicians in a room together was key to getting the vibe he wanted. To this end, he has made use of recording spaces including London's MasterChord Studio and New Orleans' Marigny Studio. 'Ode To NOLA', however, along with many other songs on the album, was tracked in his own home studio. This is a very well-equipped affair, with lots of tasty mics and outboard feeding a Focusrite Red 8Pre interface, but the live room and control room are both pretty compact. Not a problem for recording solo instruments, but enough to introduce some challenges when you want to capture a band live including a full drum kit.

For the initial live session, Robin had sensibly banished bassist Louis Thorne to the control room in search of a spill-free upright bass recording, while he himself sang and played Rhodes piano in the live room, with drummer Claire Brock on a full drum kit very close by. Electric guitarist Neil Cowlan and Pinstripe Suit brass players Stacey Dawson (sax) and Sam Sankey (trombone) added their parts later in the same studio, and the whole thing was topped off with a Hammond overdub by Robin courtesy of his Nord Electro 5D, amped through a Vox AC30 and captured by Coles 4038 and Shure SM57 mics.

When we spoke, Robin identified a couple of things that were proving troublesome at the mix, and perhaps contributing to a sense that the track as a whole was lacking in presence. Both the Rhodes and the electric guitar had been recorded through wah pedals, and whilst both sounds were very cool in isolation, they were treading on each other's toes. The upright bass sound was impressive on its own, but its powerful low end was proving hard to handle in context. Finally, Robin's live vocal mic had inevitably picked up a lot of drum spill. The Electro-Voice RE20 suited his voice very well, and has cleaner off-axis sound than most dynamic mics, but because the spill was being bounced around in a small space before arriving off-axis on the mic, it didn't sound great. Pushing up the vocal fader thus compromised the otherwise well-recorded drum sound, and adding compression or high-frequency boost to the vocal exacerbated the problem.

Small Is Booty-ful

A "lack of presence" in a mix usually means a recessed upper midrange, often paired with over-abundant low mids. This is an ever-present risk with material tracked in a small studio. To minimise the pickup of room sound and spill, you inevitably mic things quite close, with directional mics, and this brings the proximity effect into play. In this case, for example, the bass had been captured by an AEA R84A ribbon mic, and the figure-8 pattern had introduced a lot of bass tip-up. The brass overdubs were close-miked with dynamic and ribbon mics, so the musicians could play in the room together and minimise spill, but likewise the sound was a bit too warm to cut through in a busy track.

There was, therefore, a sense in which the multitrack 'wanted' to sound soft. The close-miked bass, brass and wah guitar were naturally rich in lows and low mids, while any attempt to push forward the midrange brought out the worst in the drum spill on the vocal track.

Divide & Conquer

The obvious fix for the drum spill would have been to re-record the vocal, and Robin has plenty of top-end mics with which to do it. However, the live take had that all-important vibe, which would have been hard to recapture later. Ditching the live vocal would also have meant discarding Robin's meticulously recorded video footage from the original session, which formed an important part of his promotional plans.

Translation Isn't Everything

Auditioning mixes on consumer devices such as phones, TVs, car stereos and earbuds is often recommended as a way of highlighting balance issues we might not have noticed on our main speakers. Robin had done this extensively with his own mix, and it had helped him pick up many potential issues. However, it's important to point out that listening on consumer devices is most useful as part of a broader checking regime. What's more important is to carry out level-matched A/B comparisons against suitable reference material on your primary monitoring system. Listen to your mixes on phones and boomboxes only once you're happy that the overall timbre and balance is in the ballpark, and don't get fixated on translation to the point where you start second-guessing yourself. A mix that contains nothing below 200Hz or above 3kHz would no doubt translate perfectly to any system, but it probably wouldn't be a good mix. Conversely, if your track has too much going on at 50Hz, referencing on devices that roll off at 150Hz is unlikely to reveal it.

Fortunately, help was at hand. I used to think of source separation as a technology in search of an application, but having discovered that it can separate wanted audio from spill, I'm a convert. I dropped Robin's vocal track into Hit 'n Mix's RipX, and a few minutes later, was rewarded with impressively clean vocal and 'drum' tracks. If you don't change anything, these separated tracks recombine perfectly to recreate the original; the further you depart from this, for example by reducing the level of the spill track, or processing the vocal track, the more you risk artifacts being audible. In this case, the separation was good enough that I could lower the spill fader by 7 or 8 dB, and also EQ it to remove the peaky upper-mid and muddy lower-mid frequencies that were clouding the mix. I was also able to send to reverbs and delays just from the clean vocal track, meaning there was no risk of spill feeding into those effects.

With the unwanted parts of the drum sound reduced to a manageable level, I started work on the actual drum tracks. These comprised a spaced pair of Calrec 1050 pencil mics as overheads, a beyerdynamic M88 on snare, inside and outside kick drum mics (Shure SM52a and Neumann U47 FET respectively) and a Shure SM81 on the hi-hat. There being plenty of hi-hat in the other mics, this essentially gave me four useful tracks to work with.

Claire's playing was first-rate, the kit properly tuned and the mic placements well chosen, so relatively little mix housekeeping was required to arrive at a solid basic sound. In this case, that meant time-aligning everything, bussing the two kick mics together and using Sound Radix's Drum Leveler to control »

Louis Thorne's upright bass was tracked in the control room, using an AEA R84A ribbon mic.

Thanks to Hit 'n Mix's RipX, I was able to separate the drum spill from the wanted vocal. The spill track (top) couldn't be muted altogether without introducing noticeable artifacts, but it could be reduced to a manageable level.

» their dynamics, notching out a ring on the snare mic with FabFilter's Pro-Q 3 and employing some basic EQ to tighten up the low end.

For this particular song, however, I felt the need to aim higher. 'Ode To NOLA' was supposed to be a rollicking, swaggering, tribute to one of the world's great musical cities, and it demanded more than a solid basic sound to drive it along.

Making Room

There are many ways you can add excitement to a drum recording, most of which involve compression or saturation. However, the effect of compression on drums has a lot to do with the space in which they're recorded. Compression can make the room sound suck and breathe in a way that we perceive as exciting, because it somewhat mimics the natural response of our ears to hearing very loud drumming. In this case, though, the drums had been close-miked in a small space; there was no room mic, and if there had been, it probably wouldn't have been useful.

Consequently, I decided to fake it. I'm a big fan of the 'LA Studio Drum Rooms' presets in EastWest's QL Spaces II convolution reverb: a selection of short reverb patches that enhance and blend with dry drums better than anything else I've heard. I set one of these up on a bus, fed it both from the snare track and the overheads, and routed it along with the drum tracks themselves to a single global drum bus, where I applied compression. And not just any compression: McDSP's APB system includes a plug-in called Chickenhead that brings instant attitude to almost anything. Sometimes it can be too much, but here, the combination of studio ambience and aggressive compression gave the drums the X factor they had been missing, and completely obscured any lingering trace of small room-itis.

Sending to a suitable reverb, returning it to the drum bus and then applying compression helped to add energy and life to the drum sound.

Bubbling Under

Heard in isolation, the upright bass sounded really good, thanks to Louis' excellent playing. However, this proved to be one of those occasions where achieving a sound that worked in the mix meant making it sound worse in solo! The problem was that if I set the bass fader at about the right level to fill out the low end of the mix, it was only the low frequencies that were audible, and the energy of the playing didn't come through.

I set out to rebalance the tone, using FabFilter's Pro-Q3 equaliser and Pro-MB multiband to apply some fairly drastic attenuation below 250Hz and a boost further up the midrange. This helped to rediscover the vitality and bring through the woody quality that is so characteristic of the instrument. I was then able to make the double bass much louder in the mix without overloading the low end, which is what I'd wanted to achieve. But this brought to light another issue. The transient snap of string against fingerboard had been audible even at the lower level, and now it was actually louder than the snare. It was thoroughly distracting, and undermined the nice drum sound I'd just crafted, so »

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Two instances of FabFilter's Pro-MB dynamic EQ at work on the upright bass. The instance on the left is dealing with the string snaps, briefly attenuating the midrange and high end in response to a high-frequency trigger. The other is controlling the relative levels of the midrange and bass.

I decided it had to go. I used a second instance of Pro-MB, set up to duck the mid and high frequencies in response to a high-frequency transient, along with Oeksound's excellent Spiff transient shaper. The result was a double bass sound that is unnatural in isolation, but works much better in the mix.

Fortunes Of Wah

An unusual aspect of the musical arrangement on 'Ode To NOLA' was that both the Rhodes piano and the electric guitar were played through wah pedals for most of the track. A wah pedal creates big resonant peaks which move up and down the frequency spectrum, and can be challenging to mix. One minute the instrument is pumping out a wave of mud at 300Hz; the next, it's drilling holes in your speaker cones at 2kHz. Two wahs going simultaneously means double the fun.

Thankfully, both Robin and Neil had operated their respective squelch machines with restraint, but even so, some work was required to keep both at a consistent level in the mix and avoid cluttering up the low mids. I used SoundToys' Filter Freak plug-in to trim away unnecessary high and low end on the guitar, and Sound Radix's Drum Leveler (which, despite the name, works well on all sorts of percussive and dynamic sources) to even out its dynamics. On the Rhodes piano, I did something similar with FabFilter's Pro-MB. By emphasising slightly different frequency ranges in each case I was able to make the two instruments work together rather than fight each other. Since both were played all the way through the song, it also seemed natural to hard-pan them to opposite sides, and this instantly made for a nice wide mix.

The guitar solo in the closing section of the song was particularly difficult to sit in the track. With the fader at any fixed level, some notes jumped out whilst others were inaudible. Aggressive dynamic EQ made the tone more consistent, but even then, it was necessary to do extensive level automation to achieve a relatively even sound, a process which took several mix revisions to get right. In this situation, it's often a good idea to cut out the solo section and place it on its own track, so it can be treated independently.

> "I used to think of source separation as a technology in search of an application, but having discovered that it can separate wanted audio from spill, I'm a convert."

Unexpected Delays

Once disentangled from drum spill, Robin's lead vocal required very little work. I didn't want to slam it too hard with compression lest the artifacts of the source separation become too obvious, so instead I trimmed the peaks using the analogue El Moo limiter in the McDSP APB system — most limiters are deliberately designed to be transparent, but this one adds a little more character. I then used automation to do the bulk of the vocal levelling.

Choosing vocal effects is always an interesting process, and I'm constantly surprised by how treatments that work perfectly on one voice in one track sound totally out of place in another. I had been wondering whether the vocal reverb in the original mixes was somehow contributing to the sense of things lacking presence; at any rate, I was determined that mine needed to support Robin's performance without making it sound distant. As is often the case, it seemed easiest to reach this goal by using several effects in parallel. When it comes to vocal reverb, pre-delay is often the crucial factor, and in this case I used a heavily filtered 77ms delay in SoundToys' Primal Tap to feed a short plate sound from Arturia's Plate 140. Augmenting this was a more conventional stereo slapback delay from Wavesfactory's Echo Cat, and a slightly longer treated delay which I snuck in during the breakdown.

The brass, as previously mentioned, needed some EQ to tame the low mids and, in the case of the trombone, to push the upper mids forward a bit.

I also routed both parts to a stereo bus and compressed them as one using IK Multimedia's White 2A, which helped to reinforce the sense that the section was operating as a single unit — not that they needed a lot of help in that department, as the playing was extremely tight. The brass also got its own plate reverb, with slightly different settings from the vocal 'verb.

Bus Strikes

Like many mixers, I'll often introduce master bus processing at a fairly early stage of proceedings, but it's always liable to change. I often find that a master EQ setting works well on everything apart from the drums, and that was definitely the case here. Between them, I had Arturia's EQ Sitral and Audify's RZ062A set to add a fair bit of upper midrange and high end, and driving the latter a little introduced some nice colour, but it was all getting too much on the cymbals. What I do in this situation is create two or more sub-master auxiliaries so that I can separate out the things that benefit from EQ and those that don't. These are then recombined at the actual master bus for — in this case — saturation from Goodhertz's Tupe plug-in, some low-mid trimming from FabFilter's Pro-Q 3, and compression.

On rhythm-led music like this, where kick and snare are typically the loudest sources, I find it helpful to think of master bus compression as doing one of two things. With a slow attack and fast release, the front end of each drum hit jumps out before the compressor acts, recovering again in time to repeat it for the next hit. This sort of compression thus pulls the drums out of the mix a little. Alternatively, you can dial the attack time right down so that the compressor pushes the drums back into the mix. It's not always easy to predict which approach will be most effective, but in this case, the second was clearly better. I loved the 'glue' that Overloud's GEM Comp G — an emulation of the notorious SSL G-series compressor — added with the attack at its fastest setting.

Bus compression from Overloud's GEM Comp G helped to 'glue' everything together, with the fastest possible attack setting serving to push the drums back into the mix slightly.

Down The Line

Choices we make at the recording stage can have consequences that are only felt later on in the production process. There were many good reasons behind Robin's decision to track at home; it was a space where his musicians felt comfortable, he was set up to video everything, and he has great kit. But working around the space constraints shaped the sound of the resulting recordings in a way that perhaps wouldn't have happened in a larger live room. The decision to be his own tracking and mixing engineer likewise had obvious benefits, but it also meant I was the first person to hear it who didn't have a personal investment in the project.

Referencing and mix checking (see box) are invaluable, and there are now many tools that should help identify issues with a mix. But whether it's a trusted friend, a mastering engineer or the collective wisdom of the SOS Forum, there's still no substitute for a fresh pair of ears. We all need the reassurance of a safety net, and until AI algorithms and plug-in presets can provide that, they'll never replace human beings.

Rescued This Month

Robin Phillips is first and foremost a jazz singer and pianist. He is pianist and keyboard player for jazz-house artist Berlioz, leads the Pinstripe Suit speakeasy swing band and has a successful side hustle as a piano-bar entertainer, as well as teaching piano, vocal technique and jazz theory. He has a recording studio at home where he records his own projects and those of varied clients, often incorporating multi-camera video shoots. His award-winning documentary film *Back To The Source* (https://youtu.be/sDSadbKwVgU) chronicles a road trip from Chicago to New Orleans in search of the origins of jazz and the blues. Robin lives near Cambridge with his wife, children, and dog Willow.

W www.robinphillips.co.uk

Audio Examples

To hear audio examples illustrating some of the points made in this month's Mix Rescue article, point your browser at https://sosm.ag/mix-rescue-0324. Meanwhile, 'Ode To NOLA' has now been released as a single with accompanying video, which you can watch on Robin's channel at www.youtube.com/repmusic.

SOLVING USB PROBLEMS

On Windows PCs

PETE GARDNER

The Universal Serial Bus — USB for short — may have become the most ubiquitous connection standard found in everyday use. Since its creation almost three decades ago, and over four generations, it has been implemented within many electronic devices we take for granted in everyday life. Each successive generation has increased bandwidth and, in many cases, power delivery. At the same time, they have all been designed to offer backwards compatibility with previous generations, making USB devices from different eras as interchangeable as possible (at least on paper). However, things can still go wrong, and, depending on the situation, there are various steps you can take to help rectify the problem.

Let us take a look at the different ways we can approach USB issues on Windows machines.

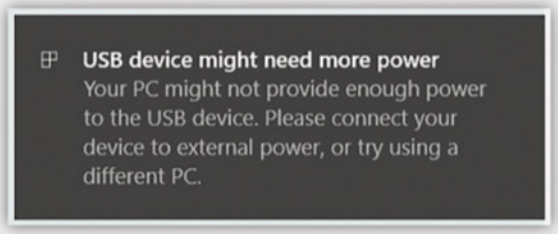
Starting with the hardware connection, the worst-case scenario is that you plug in a USB device and the PC switches off or subsequently refuses to switch on. In this case, you should check for physical damage. Look inside the USB port itself, as this symptom may be the result of the pins inside the socket being bent out of alignment, causing the system to short out when the USB cable pushes them further together.

If you plug the device into a known working USB port and the system fails to detect that you have done so (no new item appears within the Windows Device Manager), then we would tend to expect either a connection or physical issue which may lie outside of the system.

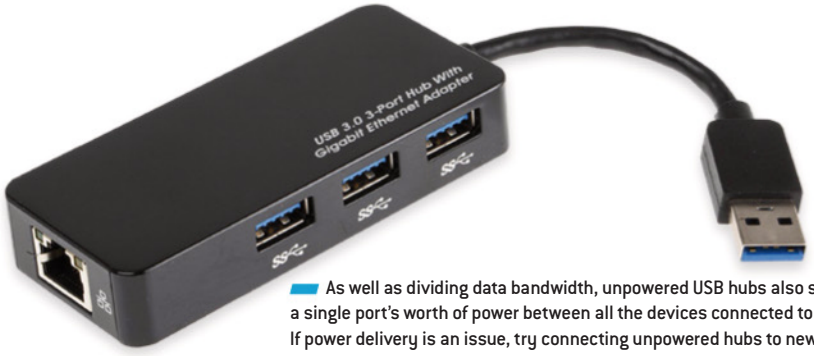
The first step would be to try an alternative USB port, preferably switching from the rear to the front (or *vice versa*) of the computer where possible. The rear ports are generally native to the system chipset, whilst the front ports tend to be supplied via third-party controllers, often resulting in differing behaviour between the two sets. You may also find a couple of USB 2.0 ports on the rear of the system, as many mainboards still ship with a dedicated pair, to help maintain support for particularly picky older hardware.

Power Move

Switching out the USB cable itself, and testing on a second computer, are both steps that could help you to narrow down where the problem lies. Upon connection, if you receive a Windows warning stating “unknown USB device



If an error appears to tell you that your USB device isn’t receiving enough power, be sure you’ve connected its power supply.



As well as dividing data bandwidth, unpowered USB hubs also share only a single port's worth of power between all the devices connected to them. If power delivery is an issue, try connecting unpowered hubs to newer-gen USB ports on your computer, as these tend to have increased power capacity.

needs more power than the port can supply" and the device has the option to add its own power source, then now is the ideal time to double-check that it is connected and powering the unit as required. Power delivery is also a good reason to avoid USB hubs without their own dedicated power source, as USB hubs powered solely by the PC connection will be limited to dividing up one port's worth of power across however many hub-connected devices are in use. However, if you only have an unpowered hub to work with, it's worth noting that newer USB generations offer improved power delivery. For example, the USB 2.0 specification allows hosts to deliver a total power output of 2.5 Watts, whereas USB 3.0 and 3.1 allow for a total of 4.5W, and the latest USB 4.0 standard increases this to a potential 240W delivery when EPR or Extended Power Range functionality is implemented. So, simply switching over to a newer-generation port may help resolve issues with power delivery.

Cable length should also be a consideration. To achieve the best performance, both USB 1 and 2 had recommended maximum cable lengths of 5m. This dropped to 3m with USB 3.0, and USB 4.0 lowered the limit again, with the older Gen 2 (20Gbps) running at its best with cables up to 2m and the latest Gen 3 (40Gbps) implementation listing just under a meter at 80cm.

This isn't to say a longer cable won't work, but you can expect a potential degradation in the speed of the transfer and an overall loss of performance. Cables longer than the recommended length may not present a problem for some less demanding devices, but if you're experiencing problems, you should make sure to check the charging wattage rating and supported USB data rate when troubleshooting. A longer-than-advised USB-C charging cable, for instance, may still provide enough data capability to allow for it to function as a USB 2.0-grade connection, offering a 480Mbps transfer rate and more restricted feature set. If you have the need to go further beyond the advised lengths then boosted active cables, or adaptors to convert the signal to run over optical or Ethernet cables, are common solutions to ensure you have stable data transfer over longer distances.

Drivers

By this point, the USB device you're troubleshooting should be showing up in the Windows Device Manager. If this is still not the case, the next step would be to check over your USB controller drivers. Windows, when first installed, includes many of the common component drivers and these can prove handy for getting up and running quickly. After installation, Windows Update »

Optimising Your USB Layout

Take a look at the connection choices on any modern PC and you'll likely find an assortment of differing USB ports on offer. The classic flat rectangular USB Type-A port has been around right from the start, from USB 1.0 through to the introduction of USB 3.2 almost two decades later. Whilst not a requirement of the standard, it's fairly common to see these ports with colour-coded innards to help denote the level of USB supported by the socket.

The original white USB 1.0 sockets were only found for a couple of years in the late '90s, being superseded in the year 2000 by the black USB 2.0 port, which remains the oldest USB standard still found on new systems today. With a transfer rate of 480Mbps and the lowest power delivery rating, it makes sense to use these ports for relatively undemanding hardware. The bandwidth on offer will be more than adequate for most simple devices, but a total power delivery of 2.5W best fits your least power-hungry devices. Your computer keyboard and mouse are obvious candidates as well as devices like security dongles, basic MIDI devices like trigger pads and smaller keyboards, or smaller audio interfaces. Many larger and more feature-rich devices may still prove relatively untaxing for the data transfer capabilities of USB 2.0, although in those instances an additional PSU may be required to power the device.

Not all USB ports are equal! Different generations of USB port are often colour-coded, with black showing USB 2.0, blue denoting USB 3.0, and red indicating USB 3.1 Gen2.

USB 3.0 ports may be indicated by blue innards, along with teal-coloured USB 3.1 Gen 1 ports. Both of these offer 5Gbps transfer rates and 4.5W of power delivery. A third variant, USB 3.1 Gen 2, is indicated by red ports and maintains the 4.5W power delivery rating but raises the transfer rate to 10Gbps. With all of the USB 3.x options, the extra power may help to provide extra functionality; for example, an audio interface might offer a better headphone amp or phantom powering on more of its inputs, whilst a well-featured MIDI controller may have its own display screen functionality. The available bandwidth and power delivery also make this a suitable choice for connecting up external SSDs or running a hub that can take advantage of the extra USB 3.0 resources.

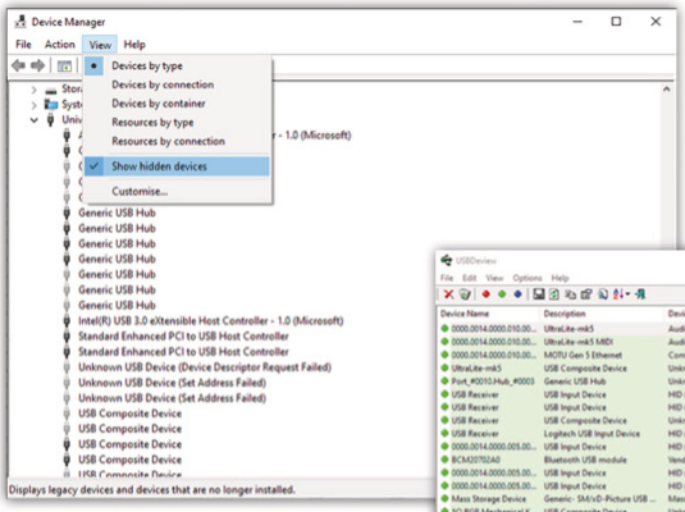
In recent years USB-C adoption has been spreading. This newer port can support various flavours of USB itself, depending upon the hardware revision. Starting initially with USB 3.1 and supporting a potential 100W, in later revisions we have seen devices capable of supporting the USB-C PD update, which raises this to 240W. USB 3.2 over USB Type-C itself comes in three revisions, from 5Gbps through 10Gpbs to 20Gpbs, along with the latest USB 4.0 standard running at either 20Gbps or 40Gbps. These increases in both power and data delivery further open up the scope to allow you to run multiple monitors directly off a single USB port, connect a hub with a selection of power-hungry or bandwidth-demanding devices, or take advantage of fast external NVMe drive storage.

Indeed, it's more general hardware like storage or display screens where the very latest USB variants tend to offer the most advantages. MIDI in itself is a very lightweight protocol, and in terms of bandwidth handling, even USB 2.0 can deliver the support required for most small recording setups. Ultimately, most devices will be rated to a given standard to support their features, and these requirements should be observed to ensure full functionality.

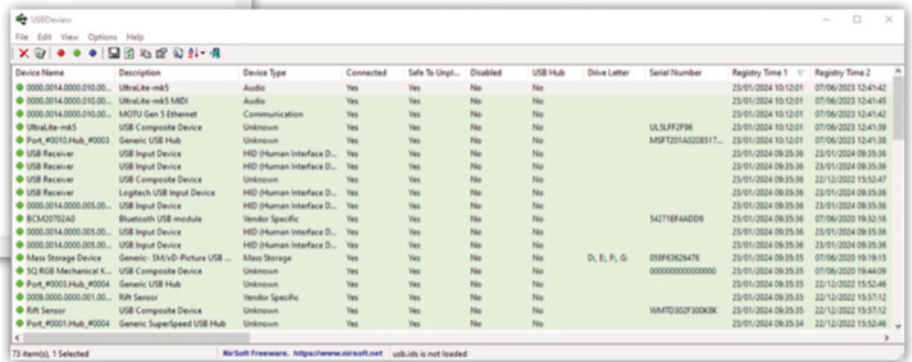
» Service may then update these further, bringing them up to the latest WHQL Microsoft-approved release, but you may find even newer drivers from other sources that include further bug fixes and updates. The first port of call is often the support pages of your motherboard supplier, where the USB drivers tend to be included within the chipset driver package; if there is a secondary USB controller on the mainboard, you will be able to quickly identify this by looking through the drivers on offer.

Although they are a good starting point, the drivers found here may still not be the most recent. If updating to the most recently available from the board manufacturer fails to resolve the issue, the very latest builds will be available from the chipset supplier, normally available directly from either Intel or AMD's own websites depending on which platform your system is built around.

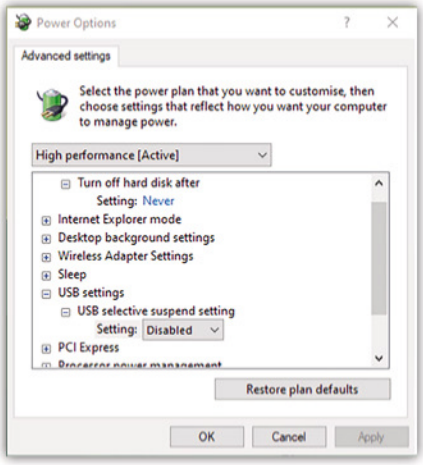
You may find that the controller is showing up within the Device Manager, but displaying an exclamation mark and noting that the device cannot start due to conflict or lack of resources. Alternatively, you may be seeing devices randomly disconnecting in use as the system reassigns internal resources. Windows treats each USB port as



When you connect a USB device to a new port, Windows will duplicate the existing driver, and these duplicates can sometimes interfere with proper operation. To see the duplicates, go to Device Manager and select 'Show hidden devices'.



The freeware USBDeview shows in-depth information on all of your connected USB devices.



its own entity, and you will find that unplugging and reconnecting a USB device into a different port will cause Windows to reload the driver to support the new connection. Whilst Windows should clean up after itself, it is possible for multiple driver instances to clash and cause devices to mis-detect or lose connection.

To resolve this, you can enter Windows Device Manager, select 'Show hidden devices' and manually remove ghosted entries or other older devices that are no longer required. This can take a little digging around if done manually, and I find that the superb third-party tool USBDeview (www.nirsoft.net/utils/usb_devices_view.html) can quickly round up all the entries for you to review in a simple-to-use application. Some older drivers like the Korg MIDI Driver observe a legacy restriction whereby Windows could only handle 10 connected MIDI devices. This limit has been long since been lifted within the OS itself, but when working with certain kit that uses older drivers, ghosted device entries can stop this hardware from working. You may find you can get it working again by disconnecting all of your USB hardware (other than the most essential items like keyboard and mouse) and then running through the remaining USB entries and removing them all. Once done, you can then plug the devices back in one at a time, allowing Windows to reinstall the drivers once more from its repository whilst ensuring all of the prior entries have been cleaned out.

Power Mad

One final USB tweak can be found within your system power options.

Disabling the 'USB selective suspend' option in Power Settings will prevent Windows from attempting to power down your devices in the middle of a DAW session!

If you open up the Windows Control Panel and select Power Options, the best general advice for any audio system is to run the Windows High Performance power scheme as a starting point. However, some further tweaks can still be applied, and crucially for USB support, there is a USB Settings section with the option of 'USB selective suspend setting', which should be set to Disabled. This stops the OS from attempting to power down connected devices when power saving, avoiding any potential device re-detection issues while your DAW is running.

USB's ease of connectivity has long been a key strength, and the ability to route it even over other standards like Thunderbolt is part of the appeal. Starting with the appearance of USB 4.0 we see the inclusion of Thunderbolt 3 within the standard, along with the ability to further add Thunderbolt 4 support at the manufacturer's discretion. Support needs to be added within the BIOS, meaning full functionality is still offered on a board-by-board basis, but this does appear to be an avenue for Thunderbolt to more widely spread across the PC platform as the two standards continue to converge.

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On Specs

Part 3: Distortion

Low distortion is often a marker of quality in audio equipment. We explain how to make sense of standard distortion specifications.

JOSHUA ISRAELSOHN

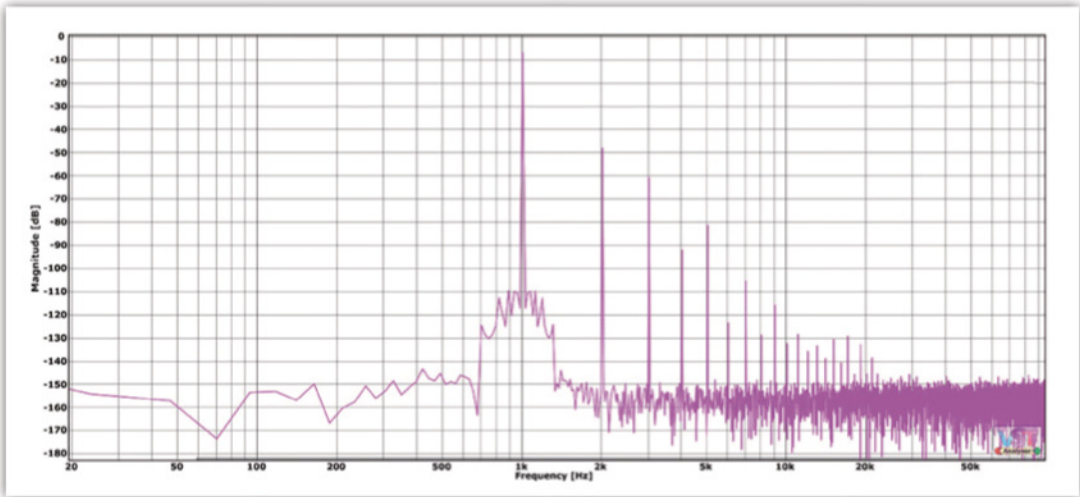
The previous instalment of this series defined distortion, for the purpose of this treatment, as "any signal component added by elements of the signal path in response to the intended audio. In the absence of an audio input, distortion is zero." Though there are others, two forms of distortion dominate in audio systems, and appear most often on equipment data sheets. These are harmonic distortion and intermodulation distortion.

Both forms of distortion derive from nonlinear electronic components — devices with outputs that are not strict multiples of their input — that necessarily form the core active (amplifying) circuits in audio-signal processing blocks. These exist throughout the analogue signal path, from a microphone transducer's or instrument's output to the analogue-to-digital converter. On the reproduction and monitoring side, similar causes of distortion exist within the digital-to-analogue converter, mixers and other signal-processing blocks, and monitor amplifiers.

Harmonic Distortion

As the name suggests, sources of harmonic distortion add harmonics — frequency multiples — of the input signal to their outputs. The presence of added harmonics can have several harmful effects, which degrade audio quality in audio capture, post-production, and reproduction signal chains when present in more than minute amounts. Harmonic distortion, for example, degrades sonic transparency due to exaggerated spectral density and, at sufficient amplitudes, reduces the perceived realism of acoustic instruments and naturally occurring sounds. At even greater distortion amplitudes, musicians put this latter trait to creative effect with many instrument pedals, such as overdrive and fuzz stompboxes, creating sounds almost unrecognisable as deriving from the raw instrument output, but here the deviation from the pure instrument tone is purposeful. In production and reproduction gear, the goal nearly always is to preserve the original signal as accurately as possible irrespective of its origin.

For the purpose of reading spec sheets, we're interested in very low levels of distortion, which are given in (fractional) percent or (negative) dB. Equipment manufacturers most often



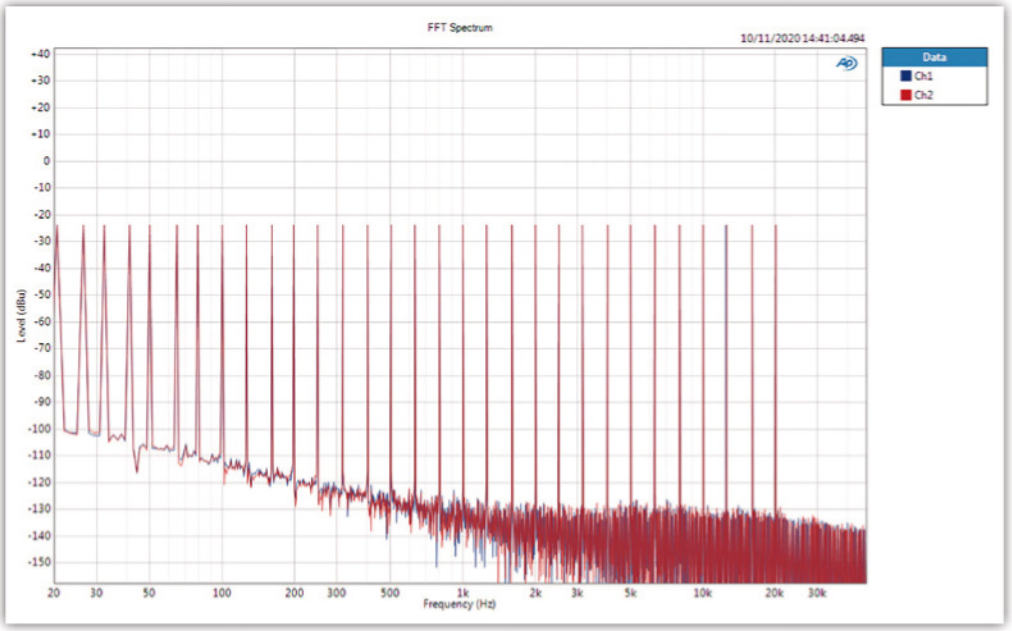
Harmonic distortion is evaluated by injecting a sine-wave signal — typically, as here, at 1kHz — and measuring the amplitude of the harmonics generated at multiples of that frequency.

state harmonic distortion as an aggregate measure of harmonics within a frequency band, which they refer to as THD (total harmonic distortion) or as THD+N (total harmonic distortion plus noise), depending on measurement method.

The measurement begins with an ultra-pure (very low distortion) sine-wave generator that provides a stimulus to the device under test. The analyser monitors the DUT's output, and filters out the stimulus frequency — the fundamental, in this exercise — through a narrow notch filter. What remains within the measurement band are the harmonic residues plus noise, which the analyser measures and presents as a fraction of the fundamental's amplitude. Analysers can apply narrow-band filters centred on the harmonics as well, eliminating the noise component of the measurement.

As we've seen with other parameters reported on spec sheets, THD and THD+N figures are only meaningful if the test conditions are stated, and these need to agree between spec sheets if device-to-device assessments are to be accurate. These include, at minimum, the test frequency and the measurement bandwidth. Most commonly, 1kHz is used for the former, which allows space for a good number of harmonics to show up within a typical measurement bandwidth of 20Hz to either 20kHz or 22kHz. But, as the BBC would put it, "other test frequencies are available", so be sure to check when making comparisons between competing products.

Note that the modest upper frequency limit of the measurement bandwidth means that you don't have to increase the test frequency much before you limit the in-band harmonics to just a few. For example, at a 1kHz test frequency, 20 harmonics fit within a 20kHz measurement bandwidth (with some attenuation possible in the last one). But at a 5kHz test frequency, three harmonics barely fit in a 20kHz measurement bandwidth, and at 7kHz you're down to one.



Intermodulation distortion is tested by injecting multiple sine waves at harmonically unrelated frequencies. These sine waves, visible here, would later be filtered out so that the amplitude of the distortion artifacts can be measured.

Intermodulation Distortion

While judicious amounts of harmonic distortion can serve musical interests, the same is not true of intermodulation distortion (IMD), which is virtually always discordant. Like harmonic distortion, intermodulation distortion results from nonlinearities in signal-processing blocks. In the case of IMD, however, the distortion signal components are the sums and differences of frequencies in the source audio.

> "While judicious amounts of harmonic distortion can serve musical interests, the same is not true of intermodulation distortion (IMD), which is virtually always discordant."

Measuring a device's IMD requires two test frequencies that are not harmonically related. A test system generates two or more ultra-pure sine waves as the input signal to the DUT. The tester's signal analysis section uses narrow notch filters to remove the test frequencies from the DUT's output signal, and measures the amplitude of the residual sum and difference frequencies. Test standards for IMD call out the test frequencies and their relative amplitudes so, when comparing two competing product's spec sheets, it's important to check that IMD measurements comply with the same standard.

Like most THD and THD+N measurements, IMD specs reflect a spot test of complex behaviour. They do not test the DUT's distortion performance across the entire audio range, which can vary particularly at the upper end of the audio spectrum. So, while comparisons of competing products' spec sheets do provide valid and valuable performance comparisons as long as their test methods and operating conditions match, they cannot tell the entire story of products' distortion performance, and they cannot predict exactly what your ears will experience under real-world audio production or reproduction applications. In other words, spec sheet comparisons serve as a crucial first-step evaluation of competing products, but they cannot entirely replace critical listening or critical thinking about what level of performance is necessary to satisfy your goals.

This series is produced in association with Audio Precision, Inc.

Now we know how to get immersive, let's add some delay and reverb to our Atmos mixes!

ROBIN VINCENT

Having covered the basics of immersive and Atmos mixing last month, I thought it might be a good idea to have a rummage through two plug-ins that are uniquely kitted out for the job: OpenAir2 and Surround Delay.

But first, I would like to go over the difference between beds and objects very quickly, because I'm not sure I fully grasped it until now, and I get the feeling that I might not be alone. A bed is a multi-channel track or bus that maps directly onto the speakers within your setup. Its outputs feed physical output channels in the same way that a stereo track goes out to a pair of outputs. When you create a surround bed, you generate a number of individual audio tracks, one for each speaker, and they are routed back to those individual and specific speakers on playback. If you are missing a speaker, then you won't hear that part of the bed.

By contrast, an object is not associated with any specific speaker or collection of speakers. It exists as a mono or stereo audio track that is mapped on playback to whatever speakers are available, based upon the panning and positional information that's created when you render the Atmos file. So, it doesn't matter how many speakers you have; an object will appear in the available speakers that are best placed to convey its position.

About Latency

Personally, when exploring effects, I like to load up a virtual instrument or a real-life instrument and play it through them. It gives you the chance to interact with things in a way that a pre-recorded sound source doesn't. However, when you're dealing with Dolby Atmos, you're required to work with rather sloth-like latency settings. At 48kHz you have to use a buffer of 512 samples, and at 96kHz you have to use 1024 samples. On my system, a 512-sample buffer setting produces a little over 10ms latency for soft synths: playable at a pinch, but some people might find it a tad laggy. For live input monitoring of sources such as guitars, that latency doubles up as the signal has to come in and go back out again, so 512 samples gives a very noticeable and probably unplayable delay.

When used as an insert, Surround Delay's taps will always follow the panning of the source track.

The alternative is to drop down to Surround Sound rather than Atmos. This gives the advantage of lower and more playable latencies, but you lose the binaural headphone monitoring, which is that marvellous thing that lets those of us without a 7.1.2 speaker array mix spatial audio. If you need to do this to achieve low-latency monitoring, all you have to do is reduce the buffer size under Options / Audio Setup / Audio Device. Studio One will cleverly disable the Atmos Renderer but leave everything else in place, so all you have to do is return the buffer size to 512 to re-engage the Renderer and continue mixing in Atmos. However, for the purpose of this workshop, I'm going to stick with Atmos and assume you're either happy with the lag or using pre-recorded source material.

Surround Delay

Let's start by setting up an Atmos project. Go to New and select the 'Mix in Surround' template. This should enable the Atmos Rendering plug-in and set you up with a default 7.1.2 speaker format. You may need to enable a second headphone monitoring output and set it to Binaural. All of this was explained in full in last month's workshop, so please refer back to that if you need to.

There are two ways to use Surround Delay: as an insert or as a send. The choice is actually very important, as it affects how it responds to the source material, its interaction with the speaker setup, and the binaural interpretation of that.

At this point, I should offer a correction to my review of Studio One 6.5, which appeared in *SOS* December 2023. In the review I stated that, oddly, the Surround Delay had no presets. Well, PreSonus have since released an update that fixes a bug that caused them to be hidden. And there's a whole bunch of them.

Create either a virtual instrument track or a stereo monitoring audio track (we'll come onto mono in a minute) and drop the Surround Delay onto the track as an insert effect. It's immediately a beautiful thing to play with, but let's see if we can better understand what's happening to make sure we can use it effectively.

By default, the track is set to the Surround Panner, so let's start here. Open the Surround Delay editor and

Here, Surround Delay is configured as a send effect, with spatial panning. The delay taps can now be panned independently of the source.

choose the '+init' preset. For the purpose of this experiment, set the Mix to 50 percent, the Level to 100, and the Beats to 1/2. This will enable us to better see the response of the delay taps coming through the bed monitoring in the Atmos Renderer.

Set the first tap (the red active one called '1') to fully right (the three o'clock position) and begin to play; ideally, use a staccato sound so you can clearly hear the delay. You'll hear the repeat on the right in your headphones. You can also see the placement of that sound by watching the bed metering. Open the Surround Panner and pan the track around while continuing to strike a key. What you should experience is that the delay tap always stays to the right of the instrument's position (depending on the width of the stereo spread), which is not necessarily to the right on your headphones. The position is always relative, not fixed. You can change the obviousness of this effect by expanding or contracting the width of the track's stereo field. If you are using a mono sound source, such as guitar, you'll find that the Surround Delay will only return taps to the same position where your instrument is panned.

So, as an insert effect the Surround Delay can only place taps within the stereo field of the source. This is because the plug-in is expecting to use a surround output and we've inserted it on a mono or stereo channel. If you look at the GUI for the effect you'll see the single speaker (mono) or pair of speakers (stereo) in the main display. So, is it wrong to use Surround Delay as an insert on a mono or stereo channel? No, because it's an interesting delay with very configurable taps that are good in any situation. However, it is designed to be used in surround...

Sends & Sensibility

For the Surround Delay to access all of our immersive speaker potential, it has to be loaded as a send effect. Do that, and you'll find that the output of the Surround Delay can be placed anywhere in the 3D soundfield. As you play, you'll notice that the metering on the FX track is showing the individual beds. If you switch the panning of your instrument track to Spatial, this will become even more obvious as the source audio is now an object rather than part of the bed structure.

Let's be a little bit more daring this time and add two taps to our Surround Delay. Load the '+init' preset again, reduce the delay time and add tap 1 to the four o'clock position and tap 2 to the eight o'clock position. As you play, you'll be able to hear those two taps in your headphones to the right and left. If you move the panning in the Object Panner, you should be able to hear that the source is changing but the delay taps are not. You can also see, now that the source is no longer being metered in the beds, that the bed metering remains the same regardless of where the source is panned.

This means that with the Surround Delay as a send effect, you can place delay taps exactly where you want them to be in the space. You could go further and place each delay tap as an Object within the Atmos space, but then you would have to set the output of the FX channel to mono or stereo and create a new channel for each tap you wanted to place. That can get very complex and unwieldy very quickly, which is why the Surround Delay is brilliant as designed. Either way, it's a lot of fun.

Now we understand where to use the Surround Delay, let's check out the features. Ultimately, you have a chain of eight delays, which you can configure independently. For each tap you can set its position in the chain, its placement in the surround space (both direction and elevation), level and feedback. The controls along the bottom for Mix, EQ and Time are applied globally.

Along the top are a few useful buttons. The Snap button turns on a magnetic pull to the speakers so that your tap placement snaps to one speaker. This enables perfect delay placement. The other buttons, Level, Direction and Elevation, change the display to show positional information for each delay.

Like Surround Panner, OpenAir2 works best when used as an auxiliary effect.

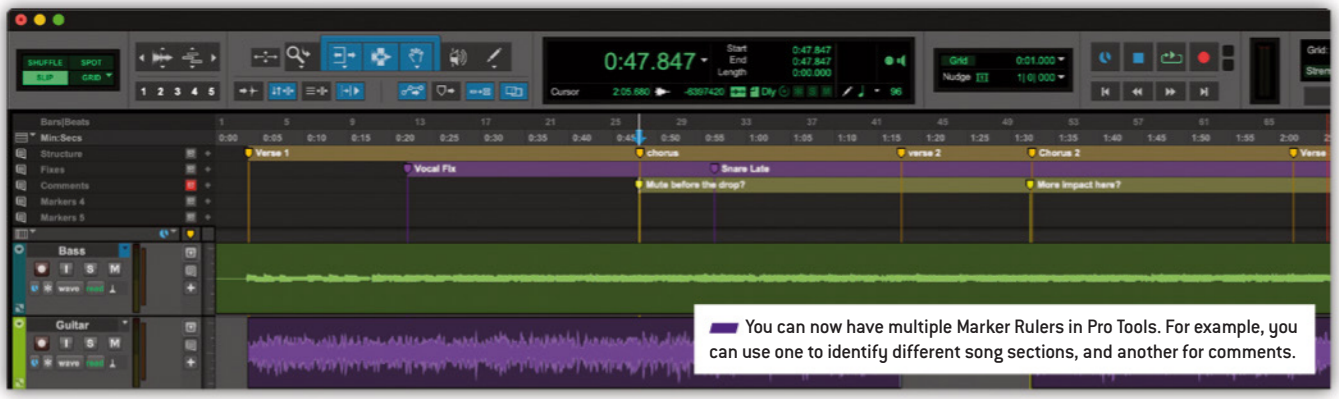
OpenAir2

Is it the same situation for the three-dimensional spaces within the OpenAir2 reverb? Yes, it is. If you drop OpenAir2 onto your track as an insert, it will follow the stereo placement of the source around your surround sound space. With a reverb, that sounds very strange, almost as though you're standing outside the room in which the instrument is being played. To stand inside the same room, you need to load OpenAir2 on an FX channel and send your track to it.

OpenAir2 comes with a good selection of immersive presets that you'll find under the 3D-IR folder. The contrast to the standard ones is quite amazing and utterly convincing. The reverbs are generated using recorded impulse responses. You can use your own IRs for this if you wish, and we'll tackle that in a different workshop.

As with Surround Delay, as you pan your instrument track, the OpenAir2 reverb will remain stubbornly all-encompassing and directionally agnostic. However, it doesn't have to be. If you open the Surround Panner for the reverb channel, you can use the Spread parameter to bunch up all the reflection channels. Then you can position the output of the reverb in the surround sound space to better match the position of your track.

In terms of editing, OpenAir2 offers two levels of control, one being super-simple while the other provides plenty of complexity if that's your thing. On the front panel, you have a little bit of control over the size, early/late reflection mix and pre-delay. That's plenty for most of us. However, there's a whole parametric EQ in here, along with detailed editing of the impulse response. You can even set levels for the individual surround speakers to find exactly the balance of reflections you need.



You can now have multiple Marker Rulers in Pro Tools. For example, you can use one to identify different song sections, and another for comments.

The new Markers and Memory Locations system makes session navigation easier than ever.

JULIAN RODGERS

Pro Tools is a very mature product, and unlike some other DAWs it is very careful to maintain backwards compatibility, but sometimes that means it can lag behind its competitors when it comes to new features. That said, I've always been of the opinion that, while Avid aren't always first, when they introduce new features they usually do a very thorough job.

One of the areas which has seen the most progress in the last 12 months is the system of Markers. Pro Tools 2023.6 introduced Track Markers, which I covered in this column in *SOS* August 2023. These are very useful, and the implementation is well thought through, but the marker-related feature I have been wanting for years is additional Marker Rulers. That is exactly what was delivered in Pro Tools 2023.12, addressing a longstanding workflow issue for me: the inability to keep markers created for different purposes separate from each other.

Markers and Memory Locations are central to managing and navigating a Pro Tools session. From quickly showing and hiding groups of tracks through to saving edit selections and recalling Window Configurations, Memory Locations are very powerful. But navigating sections of a song is probably the most popular use for Memory Locations, and before the introduction of Track Markers I, like many people, used to resort to creating a track at the top of my session specifically to populate with empty Clip Groups which I could navigate through using the Tab key, as a way of getting Marker-like functionality.

Ruler Them All

But the Marker Ruler is the most appropriate place for marking points on the timeline, and being restricted to only one ruler could be frustrating. For example, in a typical session I'll mark out sections of a song and use the 'period-number-period' shortcut on the number pad to jump from Marker to Marker. However, I'll also drop markers on the fly during playback to make notes of sections which need attention, for example minor mistakes that might need fixing. Already I'll have two categories of Markers inhabiting the same ruler. The most obvious solution is to offer additional Marker Rulers, and that is what we finally have in Pro Tools 2023.12.

Up to five independent Marker Rulers can be displayed, and they can be renamed from their default Markers 1-5 to something more indicative of their content, for example 'Structure' and 'Comments'. Markers created using the Enter button can be directed to the desired ruler using the target button on each Marker Ruler. A handy tip here for users of Apple laptops is to use the Fn button to temporarily convert the Return key to an Enter key so that, instead of returning the playback cursor to the beginning of the session, hitting it creates a Marker on the fly. Clicking in a Ruler automatically targets it, and all subsequently created Markers will populate that ruler until a different one is targeted.

Some of the most immediate workflow benefits of these new Marker facilities are exploited by a couple of new shortcuts that have been introduced. You have always been able to navigate Markers either by clicking on them in the Markers window or by recalling them from the number pad using the aforementioned period-number-period shortcut. If you spend a lot of time in the Markers window you can navigate the list using the up and down arrows (very useful when used in combination with the new filtering options, which I'll get to later). But the really useful new shortcuts that changed my Markers habits immediately were Go To Next/Previous, and the unexpectedly useful Refresh Current Memory Location. I'll explain...

Shortcuts

First, the Previous/Next shortcuts. If you don't use an extended keyboard with Pro Tools, all I'll say is you're missing out and slowing yourself down — get one! Using period+plus on the number pad you can advance to the next Memory Location, and using period+minus on the number pad you can go to the previous one. You might notice I said Memory Location, not Marker. The operation of this new shortcut might be confusing if you don't open the Memory Locations window first (Command+Num 5, or Control+Num 5 on a PC). What these shortcuts do is step through the Memory Locations displayed in the Memory Locations window.

The thing to look at to clarify what is going on is the new column on the left of the Memory Locations window. It has no icon or text in the column header, and usually displays a single white square next to one of the Memory Locations. This is the Current Memory Location. It indicates the most recently used location, and the previous and next shortcuts move this up and down the list. By invoking this shortcut you enter Navigation Mode. Repeatedly pressing plus or minus will advance the current Memory Location up or down the list. Press any other key to exit Navigation Mode.

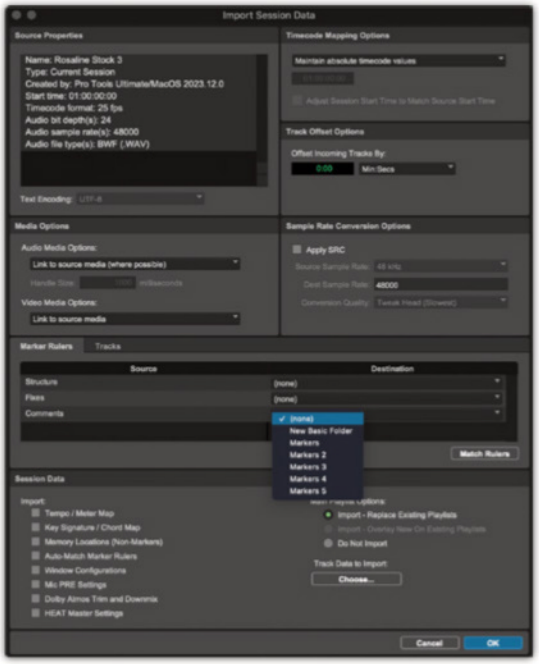
The most immediately useful new shortcut to me has been double-pressing period to Refresh Current Memory Location. When the Current Memory Location is a Marker, the playback cursor will return to the most recently used Marker location. This is particularly useful for restarting playback from the beginning of a section, especially when Insertion Follows Playback is selected, meaning that playback starts from the point at which it was stopped, like a tape machine, rather than the alternative of starting from the same point every time. Double-tapping period to restart from the last used Marker is already indispensable to me.

Filtering

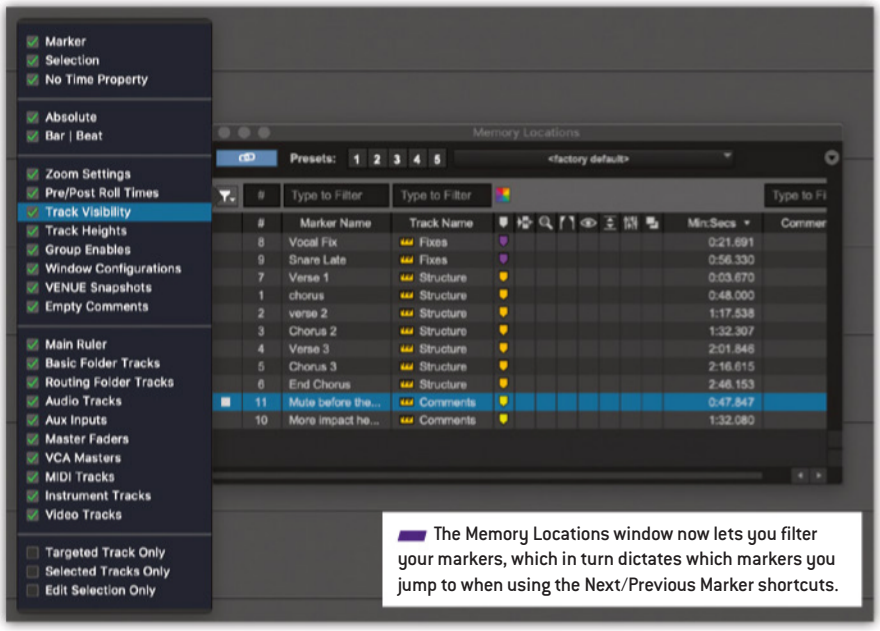
The significance of exactly which Memory Locations are displayed in the Memory Locations window and how that affects the results of using the next/previous shortcuts in Navigation Mode brings us neatly to the subject of filtering. In previous versions of Pro Tools there were filtering options, but they were essentially ways of hiding Memory Locations with particular attributes. For example, it was possible to filter out any of the properties like Zoom settings, Selection or Track Visibility by clicking on the header of the relevant columns. They worked a little like mutes, in that they made Memory Locations with those properties go away, as a mute button does for audio. The new filtering system in the updated Memory Locations window works more like a solo button, in that it hides everything apart from the property you have filtered for. The effect is cumulative, allowing complex filters to be specified which show only Memory Locations which have all of the selected properties.

There are many options including filtering by type, Marker Track, colour and by text. Filters can be stored as presets in one of the five newly added Quick Preset buttons common to other windows in Pro Tools, by Command-clicking (Control-clicking on a PC) one of the buttons. In what is a comprehensive overhaul of this window, additional functionality has been added to make managing Memory Locations easier. For example, its now possible to select multiple Memory Locations. This previously wasn't possible with Markers, because it's impossible for the playback cursor to be at more than one location. The addition of an Unlink button in the Memory Locations window means that it is possible to select a Memory Location in the list without invoking it.

Selections of multiple Memory Locations can be made by click-dragging on the list, or using Shift for contiguous and Command (Control on a PC) for non-contiguous selections. Option/Alt doesn't select all because Option invokes an eraser tool for deleting Memory Locations as it always has done, and these new features also allow batch deletion of multiple Memory Locations. It's worth knowing that Option, when paired with the up/down arrows, jumps the selection to the beginning or end of the list in the Memory Locations window.



Memory Locations now get their own section in the Import Session Data window.



The Memory Locations window now lets you filter your markers, which in turn dictates which markers you jump to when using the Next/Previous Marker shortcuts.

Import/Export

There is a huge amount more to say about these new features but something that should get a specific mention here is the attention that has been paid to handling the import of Memory Locations in the Import Session Data window. This is already a busy window, so it's good to see the addition of a new tab to cater for Memory Locations. Pro Tools now automatically scans the import for identical Memory Locations and filters out any redundant and duplicate entries. Where previous version of Pro Tools had a tick box to import 'Ruler Markers / Memory Locations', we now have a box for 'Memory Location (Non Markers)' and a whole tab handling the import of Marker Rulers. This tab allows you to specify a destination for each ruler, or the option to import them as Track Markers to a new Basic Folder track. Or if you're happy to let Pro Tools figure it out for you, there is a Match Rulers button.

This is a really useful addition to a really important part of Pro Tools. At the time of writing they are very new features and if there is an improved way to renumber Markers I haven't yet found it, but because of the new Navigation Mode I don't think I need it any more. Ideal!

Cubase 13 brought with it the welcome return of Steinberg's Vocoder plug-in...

JOHN WALDEN

For users of Cubase Pro and Artist, version 13 brought with it the return of an old favourite: Steinberg's Vocoder plug-in has finally made it into the 64-bit world and, while the basic concept remains the same, it has also undergone a smart visual makeover. Vocoders are perhaps most popular in electronic music styles, in which the classic 'robot voice' is often heard, but if you're prepared to experiment a little you'll also find that the revamped Vocoder can conjure up a much wider range of effects. In this month's column, I'll run through how you might go about this, and you'll also find some audio examples on the *SOS* website (https://sosm.ag/cubase-0324) to accompany each of the main stages I describe.

Vocoder 101

Put simply, a vocoder allows you to take some of the sonic characteristics from one sound (called the 'modulator') and apply them to another sound (known as the 'carrier') — by far the most common example is when a vocal modulator is applied to a synth-sound carrier. The pitch of the resulting sound is always determined by the MIDI note(s) used to trigger the synth, but the sound of the voice modulates the synth sound, so its character changes: the effect is like making the synth 'talk'. Depending on the MIDI note data received by the synth, you can get the classic monotonic robot voice effect or something with more melodic and/or harmonic content.

You can use sound sources other than a voice as your modulator input to Vocoder, though, and while, just like most vocoder plug-ins, Vocoder includes a synth engine to serve as the carrier, with a little side-chain tomfoolery its magic can also be applied to an another synth, such as Retrologue or Padshop.

Insert This Way Up

Vocoder an audio effect plug-in, so the most obvious options is to place it in an Insert slot on an audio track, and I'll focus on that route here. But note that you could use it as a send effect inserted on an FX track. A scenario where that might be useful is where you know that you'll want to blend an unprocessed (or differently processed) version of the modulator sound with the 'vocoded' version.

The first screen summarises the basic configuration for the Insert effect route. An instance of Vocoder has been inserted in the top-most audio track (coloured red). This contains a sung vocal melody that will act as our modulator. Vocoder's UI is shown in the middle of the screen: the Carrier section provides the controls for the internal synth engine, while the Modulator section controls how the incoming audio signal is used to modulate the carrier sound. The specific settings I've used here are based on the Smooth 16 preset but with a few of my own tweaks, and are easy to recreate. Note that MIDI is set to External (allowing you to control the pitch from a recorded MIDI track or external keyboard), the Bands parameter is set to 16, and both the Talk Thru and Gap Thru settings are at zero percent (I'll come back to these last two options).

A classic vocoder setup, with the vocal audio track (red) acting as the modulator and Vocoder's internal synth providing the carrier. In the small inset image (highlighted in the blue box) you can see in the MIDI track's Inspector panel that the MIDI out from this track has to be routed to the specific instance of Vocoder that's inserted on the audio track.

The bottom-most MIDI track (in green) provides the MIDI input to Vocoder to control the pitch of the carrier (synth) sound. As shown in the small inset image from the MIDI track's Inspector panel, the MIDI out from this track has to be routed to the specific instance of the Vocoder plug-in. If you're playing MIDI note data in 'live' during playback rather than using pre-recorded MIDI data, you'll need to select the MIDI track and record enable it (or engage the monitor button) for the note data to be forwarded to the Vocoder.

Do Adjust What's Set

Page 158 of the PDF Plugin Reference Manual takes you through all of Vocoder's controls in detail, but a few are worth

highlighting here. For example, in the Carrier section you can use the Noise Mix (and Noise Mod) and/or Bright controls to blend in a bit of an 'edge' to the processed sound if you need it to cut through a mix a little more. In the central panel, adjusting the number of frequency bands in the processing will influence the audio quality of the result, with more bands tending to allow the nature of the modulator signal to come through more clearly in the final output.

In the Modulator section, the Min Freq and Max Freq act almost as high- and low-pass filters, while the Bandwidth knob, which sets the frequency bandwidth used by each band, can dramatically change the tonality of the eventual sound (higher values produce a fuller sound).

The function of the Talk Thru and Gap Thru controls are worth noting. Both controls let you blend in the unprocessed modulator sound into the final output. Talk Thru sets the level of this unprocessed sound while Vocoder is receiving notes, and Gap Thru sets it when no MIDI notes are being received. You can, of course, set a balance of the two controls that allows the unprocessed modulator source to be heard at all times, but they give you more flexibility over when, and how much, the unprocessed modulator sound source is heard than a simple wet/dry control might.

Let's imagine that you want to hear the unprocessed vocal most of the time, but trigger the Vocoder as a spot effect so that the processed sound totally replaces the unprocessed one only on a few words/phrases. In this case, you'd set the Talk Thru control to zero and the Gap Thru control to a suitable non-zero value. On playback, in the absence of any MIDI note input, you'd hear just the unprocessed modulator signal (a vocal in this example). Then, as soon as the Vocoder received a MIDI note (or notes), the unprocessed sound would be replaced by the vocoded sound. Different combinations of these controls allow you to achieve different outcomes to suit your needs.

Take Note

While the sonic character is controlled by the nature if the carrier and modulator, the MIDI notes play a significant role in the musical usefulness of Vocoder's output. Single note lines let your synth 'sing' the phrase in a melodic fashion. If you use MIDI notes whose length spans several syllables or words of the original sung phrase, then you can easily achieve the classic (clichéd?) robot voice effect. But if you match the timing of your MIDI note onsets to those of the sung phrase, you can create all sorts of alternative melodic variations not present in the original.

You can use MIDI chords as your note input too, and in this case Vocoder will generate vocoded harmonies based on those chords. This can generate some quirky backing vocals if you just follow your project's chord progression and, depending on how you play the chords (simple block chords or with more variety by adding inversions or extensions), you can create either a static robotic style or something more akin to a real backing vocal group, though of course with a more synthetic quality to the actual sound.

Incidentally, you can feed a live audio source into your Vocoder track and play in live MIDI note data at the same time, so that whatever you sing will be 'vocoded' on the fly. Because you're performing both the modulator input and MIDI note input at the same time, it's very easy to get them in sync; simple melody or chords, there's a lot of fun to be had here.

Don't Do Normal

The examples described above use a combination of a voice-based modulator and Vocoder's synth engine as the carrier. That will let you create the classic vocoder effects, but if your creative streak runs deeper there are two further options to explore. First, you can experiment with different input sounds as your modulator. For example, vocalised vowels (rather than sung words) can be interesting, especially if you change the tonality of the sound as you sing — in effect, you're using your vocal sound as a type of sweepable band filter — and many solo instruments, notably guitars, can be used in a similar fashion. Or, if you want things to get really weird, try something rhythmic like drum or percussion loops, as I've done for some of the audio examples.

Second, you can experiment with using different synths (or other sources) as your carrier sound simply by routing them to Vocoder's side-chain input — the internal synth engine will be bypassed and the side-chain source used as the carrier. Given the basic nature of Vocoder's synth engine, you might imagine that using a more sophisticated synth such as Retrologue or Padshop would instantly create a more interesting result. It might, or it might not: picking the modulator and carrier that might play nicely together can be something of an unpredictable process that requires experimentation and a little patience — but it can be rewarding too, and well worth the time investment!

By using Vocoder's side-chain capability, you can use an external synth (in this case Retrologue) to supply the carrier sound.

1. The +/- octave delayed shimmer setup.

Dual effects chains have huge creative potential.

PAUL WHITE

I've already covered DIY shimmer reverb in this column, and with a little imagination, it is possible to create many other variations on the theme of less natural-sounding reverbs that can be put to creative use, especially when working on ambient or cinematic music. Many of these treatments rely on parallel chains of processing, and there are a couple of ways of achieving this in Logic. Perhaps the simplest is to use some or all of the plug-ins in dual-mono mode rather than stereo, as this allows you to have completely different settings for the left and right channels. If you are working on a mono track, insert the Direction Mixer first to convert its signal path to stereo.

Octave Delayed Shimmer

Let's say you want to place an eighth-note delay before one side of a reverb, but leave the other side working normally. All you need to do to achieve this is to insert a dual-mono delay before the reverb, set one channel to 100 percent wet, zero percent dry with an eighth-note delay time and then set the other channel to zero percent wet, 100 percent dry. For a single delay, set the feedback to zero. Feed this into a dual-mono reverb and you have your delay offset. Of course, your dry sound will also be affected by the delays and reverbs, so if you want to have full control over the wet/dry balance, copy the audio part to a new track and use that as your dry signal.

With this arrangement, you can set the two reverb paths identically, but you can also bring in further processing that differs between the left and right paths. For instance, try applying heavy filtering to a longer reverb on one side so that the timbre of the reverb appears to change as it decays. Another simple trick is to use a dual-mono pitch-shifter before the reverb to sharpen one side by a few cents and to flatten the other by a similar amount. This adds a useful texture to the reverb. Or you can shift one side up by an octave and the other down, with different delay times feeding each side. This stacking technique can be used to create novel stereo treatments using any effect plug-ins that offer a dual-mono mode. Again, use a copy track as your dry signal, as this allows you to set the effects fully wet. If your dry signal sounds too dry, you can also add a dash of conventional reverb to that, and if your delay sounds still come across as too distinct, you can put two reverbs in series to really diffuse the sound as shown in Screen 1. Logic's pitch-shifter is not the smoothest around, so I'd suggest you pick Manual mode and set both Delay and Crossfade times to maximum. This loosens the timing slightly, but as we're processing reverb, that doesn't really matter, and it does produce the smoothest results.

For the Screen 1 patch, I start with the Direction Mixer followed by a dual-mono tape delay. This is set to delay one side by one second and the other by 1.7 seconds, 100 percent wet — though you can always pick a tempo-sync option too. This feeds a dual-mono pitch-shifter with one side set to +12 semitones and the other to -12 semitones, though you can also experiment with fourths and fifths. Again, this is set to 100 percent wet. Next, I've used Logic's Enveloper to slow the attack of the effected sounds so that they don't come in too abruptly. After that come two SilverVerbs in series, both set to 100 percent wet to really diffuse the shifted sound. Mixed in at a modest level, this produces a very atmospheric effect.

Using a similar strategy, you could place something like a Scanner Vibrato emulation before the reverb (Rotary Speaker doesn't have a dual-mono option, and the same is true of some other Logic plug-ins), again set for dual-mono mode, with one side running fast and the other running slow. Should you want to use a different effect entirely before each channel of the reverb plug-in, simply insert both effects before the reverb but after the pitch-shifter, again in dual-mono

mode, then use the wet/dry control of the dual-mono plug-ins to essentially bypass the left channel of one and the right side of the other. (The plug-in bypass control button affects both channels in dual-mono mode, which is why you need to use the wet/dry mix control.)

Pitch Tremolo

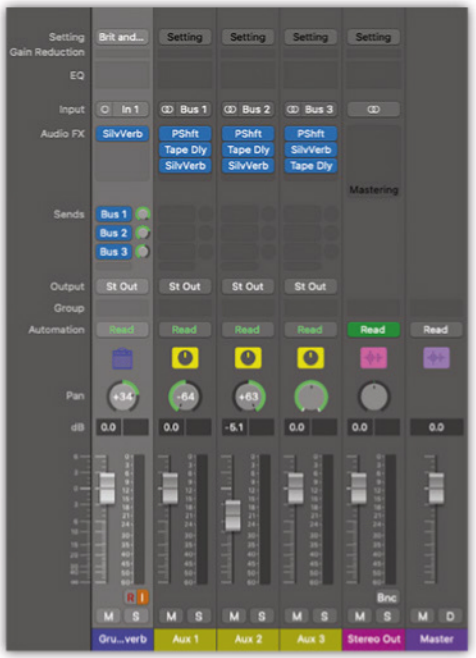
Another novel treatment I set up using a chain of dual-mono plug-ins was to place a Tremolo plug-in at the start of the chain (again using the Direction Mixer if it's a mono track), configured as a tempo-sync'ed square-wave panner. By placing different pitch-shifts on each side before the signal finally hit the reverb, I was able to create rhythmic pitch-shifts as the two differently processed channels alternated. This setup is shown in Screen 2. If you want to take the rhythmic thing to another level, Logic's Step FX can be thrown into the mix instead of a simple panner, and you can also add delays. To make the panning effect more obvious, you can put another panner at the end of the chain rather than at the start, to hard-pan the reverb sound. The best thing about this approach to multi-effect creation is that you can save the entire setup as a User Channel Strip so that it is always available, and again, you can copy the source audio to a dry track to give you scope for blending the processed and clean sounds.

Taking The Bus

Another way to create parallel effect chains is to set up bus sends from your source track, each bus feeding a different chain of effects where the reverb is usually at the end of the chain. Putting a reverb (or two) after Logic's pitch-shifters is a good way of disguising any pitch-changing artefacts, as is using the Enveloper plug-in to soften attacks. There are also occasions when placing a modulation effect after the reverb can be effective, as the strength of the modulation effect won't be diffused by the reverb.

I used the bus approach to try to emulate a rather lovely effect that I believe is called 'Into Dust', created using the Meris Mercury X pedal, where a normal reverb is joined by an octave-up or octave-down reverb after a couple of seconds or so. This is similar to the octave shimmer described earlier, but much more controllable when set up using buses. Use one post-fade send for your 'normal reverb', and a second feeding a 100 percent wet, single long delay (no feedback) followed by a pitch-shifter and another reverb. Unless you're treating drums, the low-horsepower Silververb works very well for this type of effect, and the bus fader make it easy to adjust how much of the shifted reverb comes in.

Add another send feeding a similar chain of delay, pitch and reverb, but this time add more delay to the second chain, and if the first send is producing octave-down reverb, set the second one to bring in an octave up or maybe add a musical fifth. If the onset of the octave reverb seems too fierce, you can always insert the Enveloper plug-in after the pitch-shifter and set it to its longest attack time (200ms). Now your reverb tail will morph through three distinct timbral stages as each pitch-shifted version kicks in. I got close to the effect I was aiming for; mine sounded a little different from the pedal version, but was nonetheless very usable.



3. Bussing in an attempt to recreate the Meris Mercury X 'Into Dust' patch.

This type of long, treated reverb can sound very messy on busy parts, but on sparse piano or guitar lines, it can add real magic. You can get close to this using the channel insert technique described earlier, where dual-mono reverbs are used in combination with a copied dry track, but the advantage of the bus approach is that it is much easier to adjust the bus effect levels as well as their pan positions to fine-tune the effect, and you aren't limited to just two effects pathways.

The down side of the bus approach is that you can't save everything as one User Channel Strip setting — you need to save the source track channel plus separate User bus settings for each of the bus chains that you set up. Even so, as long as you name things sensibly, it doesn't take long to call things up when you need them. If you come across a combination that you might use on a regular basic, you could include your favourite setups as part of a song template.

2. Rhythmic pitch shifts with the Tremolo effect.

DP's comp and take management features help assemble the perfect performance.

The dot after the take name (highlighted) tells you that the track contains more than one take.

MIKE LEVINE

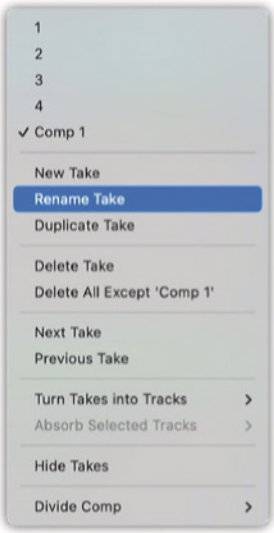
Ideally, every recording would be flawlessly executed in a single take. But the reality is that we often get the most favourable results by piecing together the best parts of multiple takes into one composite track, especially with vocals. DP users are fortunate to have a comprehensive suite of comping features that are powerful and straightforward.

What's Your Take?

Before getting into the specifics of comping, it's crucial to understand how DP handles takes, because they are the building blocks of a comp. To make a comp, you need a track that contains at least two takes.

When you add a new track, whether audio, instrument or MIDI, you'll see in the Takes menu (which you'll find in the Track Settings Panel of any track in the Sequence Editor) that it's labeled 'take 1' by default. Each time a new take is recorded, the number gets automatically incremented. You can tell if a track contains more than one take because the name in the Takes menu has a dot after it.

During the recording process, you typically create additional takes with the New Take command from the Takes menu before recording each pass. You can change a take's name with the Rename Take command from that same menu.



Rename Take is one of many useful commands in the Takes menu.

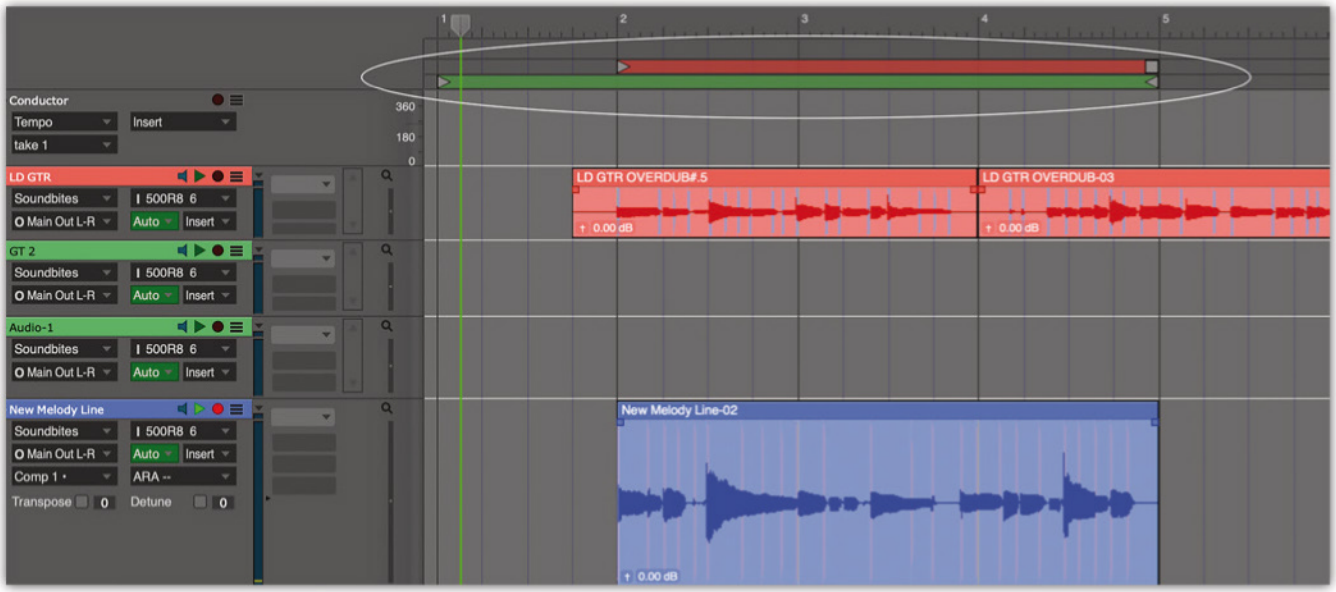
Getting Cyclical

Another common way to create multiple takes is to record them continuously using Cycle Recording, so that the singer or instrumentalist records a new version each time the Cycle loops. Recording this way works well when focusing on a finite section of a track. For example, if you want your vocalist to record one verse at a time and do multiple versions of each, you could Cycle Record with the Cycle range set for the verse you're working on.

Turn on Cycle Recording by clicking on the Memory Cycle button, which is under the transport controls. Set a time range by dragging in the Memory Cycle Strip, which shows a green line representing the range. You also need to turn on the Overdub button to make Cycle Recording work.

If you're Cycle Recording over a specific section of a track, such as one phrase of a vocal, it helps also to turn on Auto Record and set your Cycle's start point a measure or two before the Auto Record region.

The Memory Cycle and Overdub buttons are turned on to enable Cycle Recording.



The circled section shows the Memory Cycle set longer than the Auto Record region, which allows the performer time to get set before each take.

That way, you have a little time to get ready before DP starts recording again.

Absorb Tracks

When using Cycle Recording or creating new takes using the menu command, you'll record multiple takes into a single track. Occasionally, however, you might run into a situation where you want to create a comp using material recorded on separate tracks. That could happen if you're comping a performance recorded in another DAW and had to import each take into DP as a separate track. Or, possibly, you recorded in DP but didn't use the Takes feature.

The answer for such a situation is the Absorb Tracks command in the Takes menu. It lets you combine separate tracks into one track as takes. First, select the tracks you wish to absorb. Then, in the target track, choose the Absorb Tracks command, which gives two choices: Current Takes or All Takes. The former will bring in any other selected track and turn it into a take, but won't include any takes nested in the absorbed tracks. To do the opposite, you can use the Turn Takes into Tracks command, which is also in the Takes menu. It will convert each take into a separate track. You could use this feature to turn a harmony track, recorded in several passes, into a thicker, layered part. You could also use it to produce a double of a lead vocal track.

Ready To Comp

Once your takes are recorded, you can start the comping process. First, select the Show Takes command in the Takes menu. You'll then see each take in a separate lane, stacked vertically. MOTU call this the Take Grid.

Above the takes is another lane containing a newly created comp track. Notice that the take is labelled Comp 1, and the headers of the take tracks are slightly indented. Because the comp and the recorded takes are the same colour, you may want to change the Comp track to a different colour to make it stand out even more.

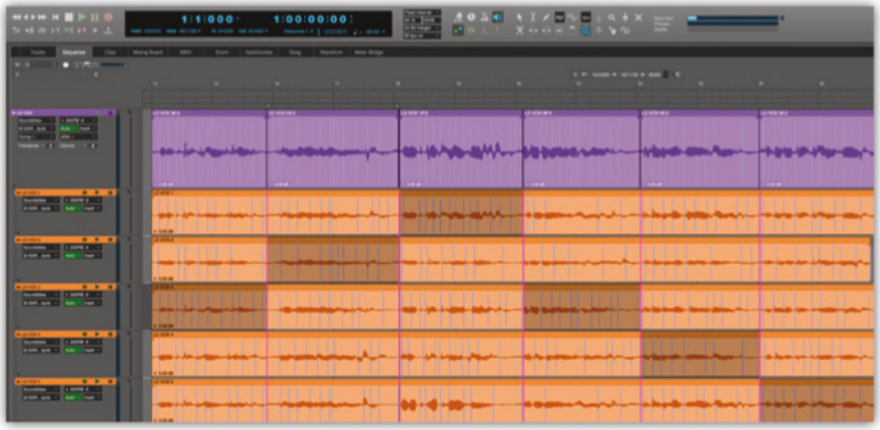
Don't be confused when you see that Comp 1 already has audio. When you first create a new comp, it will contain audio from take 1, which becomes shaded to indicate that it is currently being used in the comp.

When Show Takes has been invoked, DP's transport will play whatever is in the comp. That's a key to how the comping feature works. Using the Comp tool (which I'll explain shortly), you designate the sections you want from the takes in the Take Grid, and they will show up in the comp.

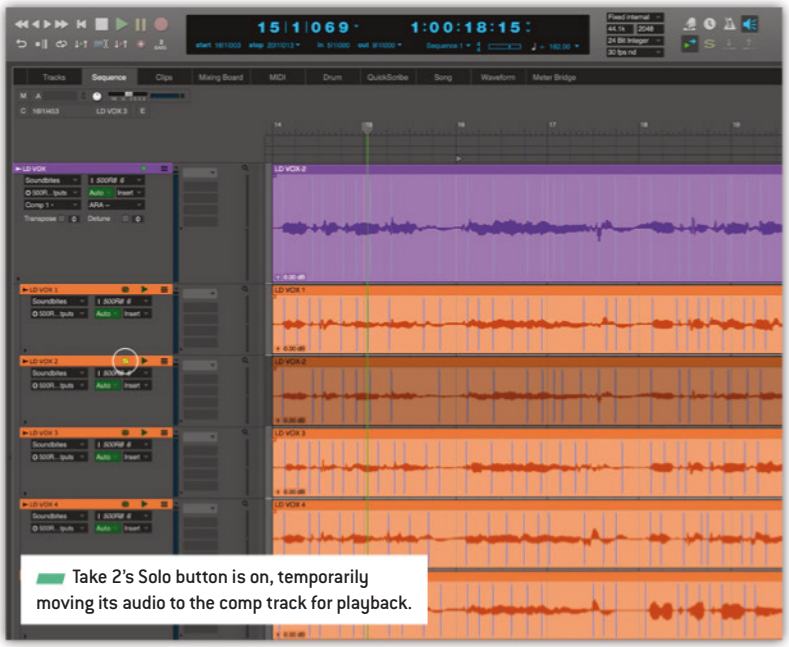
If you want to listen to a specific take, press the Solo button to the right of the take name. When you do, you'll see that the entire take is shaded, and DP has moved its content temporarily to the comp track, which makes it active for playback. When you deselect the Solo button, the comp reverts to the previous content.

Next, let's focus on the Comp tool, which resides in the Tool Palette. You can quickly invoke it by hitting the B key twice. Alternatively, you could set it as the Alternate tool and call it up by holding down the X key.

The Comp tool has a few functions, but the main one is to designate the sections in the Take Grid that will appear in the comp. You do this by clicking and »

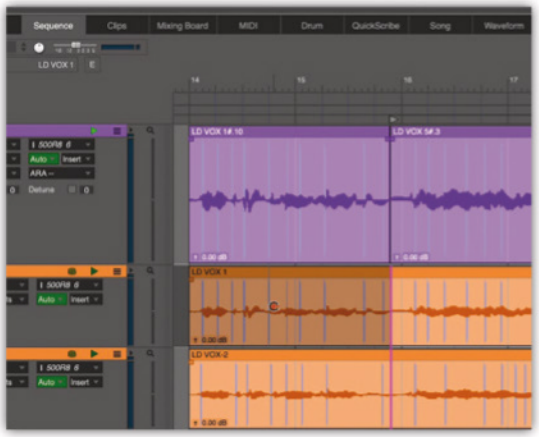


The shaded sections of the Take Grid appear in the comp at the top.



Take 2's Solo button is on, temporarily moving its audio to the comp track for playback.

The Comp tool is the one selected in the second row.



Clicking a section in Take 1 with the Comp tool.

» dragging the Comp tool over the section of the take you want to use and then releasing the mouse. You'll see that DP creates vertical red divider lines at the beginning and end of your designated region, across all takes. When you click on any take between the two dividers, that area gets shaded, and DP sends it to the comp track.

DP's grid snapping applies to the Take Grid and comp track. So if you want to make your section dividers line up with bars or beats, turn snapping on. For unconstrained selection, turn snapping off.

You can select a dividing line by clicking on it with the Comp tool, which turns it white. Once selected, you can drag it to reposition it or press Delete to remove it.

Strategic Thinking

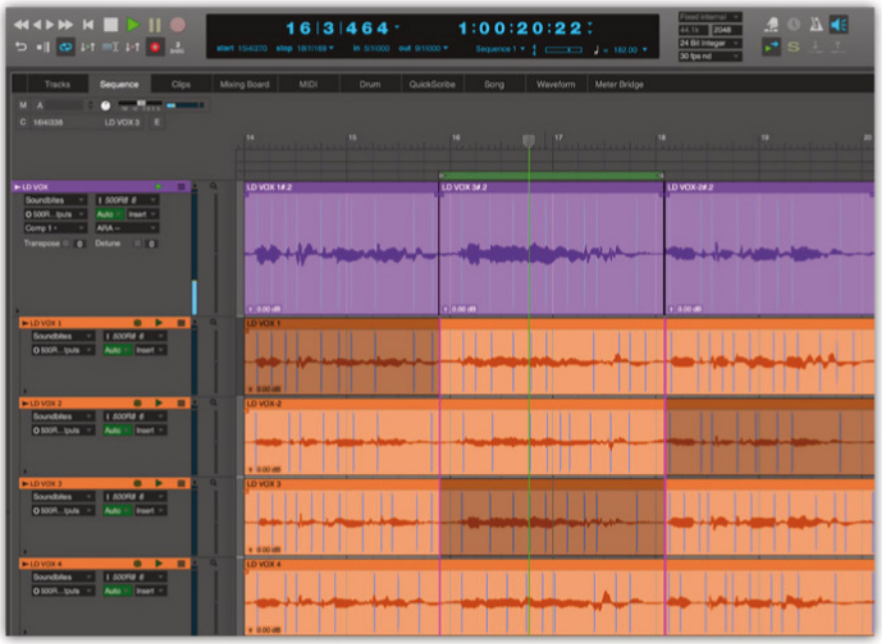
How best to divide the Take Grid will depend on the material and your work style, but for vocals, separating every phrase is an excellent place to start. That way, you can compare the various takes one line at a time.

An effective way to compare your takes involves DP's Memory Cycle feature. Start by using the Comp tool to define the section you want to concentrate on, whether a phrase, a verse or any other element. Next, turn on Memory Cycle and set its range to be the same as that section. Hit play, and as the section loops, you can listen to the various takes by clicking on them with the Comp tool.

When you've finished assembling your comp, choose Hide Takes from the Takes menu. The Take Grid will disappear, and the track will contain your comp. You may need to crossfade the various Soundbites within it and clean up any extraneous audio. (You can also edit your takes while they're in the Take Grid. The editing features are the same as in a normal DP audio track.)

If, later on, you decide you need to go back into the comp's Take Grid to make adjustments, make sure the comp is currently selected in the Take menu and then simply choose Show Takes again.

DP lets you assemble as many comps as you want. Here's how to create another comp: With Hide Takes invoked, reopen the Takes menu and select New Take. Next, choose Show Takes; your takes will appear without the dividers from the previous comp and with a blank comp track at the top. You can then freely switch between multiple comps. Like so much in DP, the comping features are deep and flexible.



Using Memory Cycle, you can easily listen to each take of a particular section.

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High-end Mixing Headphones

LUKE WOOD

Although it's safe to say the majority of engineers prefer to mix using monitors, there are plenty of situations where a high-quality pair of cans will come in handy. Checking a mix on headphones can reveal some crucial details that may have been missed when listening on speakers — even for those working in a well-treated room — and they're certainly a lot easier to carry around and set up if you need to work on a project whilst on the move! In this month's Spotlight, we take a look at a selection of headphones that have been designed to tackle serious mixing and mastering tasks.

Audeze LCD-X

AKG K712 PRO

The K712 PRO is based around a pair of carefully selected drivers that feature the company's flat-wire voice coil construction, which is said to deliver fast transients along with a detailed high-frequency response. A soft leather headband and lightweight construction help to maximise comfort over long sessions, whilst the open-back design delivers a natural, spacious sound with precise stereo imaging as well as helping to avoid listening fatigue. It also features a detachable cable which locks safely into place using a mini XLR connector.

$ $659
W www.soundonsound.com/reviews/akg-k612-k712
W www.akg.com/Headphones/Professional%20Headphones/K712PRO.html

Audeze LCD-X

Audeze's Reference Series promises to deliver a listening experience comparable to high-end speaker systems, with a neutral response and low distortion figures helping to combat listening fatigue. The best-selling LCD-X employs a number of patented technologies, including Ultra-Thin Uniforce diaphragms and Fazor waveguides, whilst efficient planar magnetic drivers offer a low impedance that the company say allow the model to deliver a great sound from almost any device with a headphone output. The physical design has been updated since its initial release, and the latest version boasts a reduction in weight along with a suspension headband design and cushioned earphones that offer improved comfort over long sessions. There's also a closed-back version in the form of the LCD-XC, as well as the LCD-MX4, which further improves the design's efficiency thanks to technology derived from the company's Flagship Series. All of the Reference Series models can be used alongside Audeze's Reveal+ plug-in, which provides personalised HRTF files and emulations of world-class studio spaces, and is available at a discounted rate when purchased with the headphones.

$ $1199
W www.soundonsound.com/reviews/audeze-lcd-x-el8
W www.audeze.com/products/lcd-x

Audeze MM-500

Developed in collaboration with Grammy-winning engineer Manny Marroquin, the MM-500 has been designed to deliver studio-quality sound anywhere whilst allowing users to create consistent mixes that translate well across other systems. Like the Reference Series offerings, it features planar magnetic drivers which offer fast transients and a neutral frequency response, along with a low impedance that allow them to be driven easily from a wide range of devices. An adjustable spring steel headband and plush earpads help to ensure a comfortable fit, and a machined aluminium construction combines a lightweight feel with durability. Audeze have recently introduced another model to the series, the MM-100, which delivers many of the same features as the flagship MM-500 but at a significantly lower price point.

$ MM-500: $1699 MM-100: $399
W www.soundonsound.com/reviews/audeze-mm-500
W www.soundonsound.com/reviews/audeze-mm-100
W www.audeze.com/products/mm-500
W www.audeze.com/products/mm-100

Audio Technica ATH-R70x

The ATH-R70x is Audio-Technica's first pair of open-back headphones, and has been designed specifically for mixing, mastering and critical listening. It promises a natural and spacious sound thanks to acoustically transparent aluminium honeycomb mesh earcup housings, as well as minimal distortion, a balanced tonality with an extended high-end frequency response and detailed transient reproduction. The physical design is very lightweight, and the R70x comes equipped with a new and improved version of the company's 3D wing support system, which promises to deliver even greater comfort over long listening sessions.

$ $349
W www.soundonsound.com/reviews/audio-technica-ath-r70x
W www.audio-technica.com/en-gb/ath-r70x

Austrian Audio The Composer

The latest addition to Austrian Audio's headphone range utilises a new Hi-X driver design featuring a diaphragm coated in a diamond-like carbon, which is said to deliver precise reproduction throughout the frequency range along with low THD levels. A comfortable fit is ensured by tiltable earcups and a lightweight design with a mesh headband, and the cables attach directly to the headband to avoid any additional strain. Balanced cable options with four-pin XLR and Pentaconn connectors are provided, along with a standard quarter-inch/mini-jack TRS cable. The company have also announced their first headphone amplifier, named the Full Score one. As you might imagine, this has been designed to pair perfectly with The Composer, and the system as a whole has been developed with high-end mixing, mastering and listening applications in mind. Full Score one is due to ship soon, although no exact date is confirmed at the time of writing.

$ The Composer: $2699
Full Score one: $1499
W austrian.audio/headphones/thecomposer/

Avantone Planar

As the name suggests, the follow-up to Avantone's Mixphones is a planar magnetic headphone design, offering an accurate sound that couples a detailed high-frequency response with natural low-end reproduction and a fast response time. Despite having a large earcup to accommodate the drivers, the Planar has been designed with comfort in mind; Avantone say that its weight is perfectly suited to long sessions, and that hours will only feel like minutes whilst wearing them! The removable cable can be plugged into either side, depending on which is more convenient, and the Planar is available in two finishes: black, or a more visually striking red.

$ $399
W www.soundonsound.com/reviews/avantone-planar
W avantonepro.com/en/products/planar-ii

» beyerdynamic DT 1990 PRO

Beyerdynamic headphones will be a familiar sight to many studio users, with models like the DT 770 and DT 150 proving to hugely popular choices for tracking duties. The DT 1990 PRO shifts the focus to mixing and mastering, promising to deliver a detailed and balanced sound with a natural stereo image thanks to the pairing of the company's Tesla driver technology and an open-back design. The physical construction has been designed not only to ensure comfort, but also to assist the audio quality — two pairs of interchangeable earpads are provided, allowing users to tailor the sound to their liking. Coiled and straight cables options are provided, and are attached via a mini XLR connector.

$ $529
W global.beyerdynamic.com/dt-1990-pro.html

Focal Clear Mg Professional

Focal's Clear Mg Professional model employs full-range drivers that sport an 'M'-shaped inverted dome, a design feature derived from the company's range of studio monitors. It is said to create an an extremely precise sound that delivers detail across the full audio spectrum whilst maintaining a flat and natural tonal balance. The headband has been designed to create a constant curve that distributes weight evenly, whilst a rotating mechanism and memory-foam earcups assist sealing for optimum low-end performance and ensure a comfortable fit. Focal say that the headphones will sound their best after a 'running-in' period, and recommend at least 24 hours of bass-heavy music at a relatively high playback level to stabilise the drivers.

Focal Clear Mg Professional

$ $1499
W www.soundonsound.com/reviews/focal-clear-mg-professional
W www.focal.com/uk/monitoring-speakers/professional-headphones/clear-mg-professional

HEDD Audio HEDDphone TWO

Air Motion Transformers, or 'ribbon tweeters', can be found in a number of speaker designs, but HEDD Audio were the first company to employ them in a pair of headphones. Now, with the release of the HEDDphone Two, they've managed to reduce the size and weight of the original design whilst maintaining the impressive technical performance. The AMT driver delivers fast transients and an extended high-frequency response, whilst a technique called Variable Velocity Transformation allows the folds in the driver material to vary in width and depth, resulting in an even frequency response across the audio spectrum. The new design also features a revised headband design (named the HEDDband), which offers adjustment over both its height, width, curvature and tension to ensure a comfortable fit.

$ $1999
W www.soundonsound.com/reviews/hedd-audio-heddphone-two
W hedd.audio/products/heddphone-two

Neumann NDH 30

Neumann say that their aim when developing the NDH 30 was to make the sound of their KH monitors, calibrated with their MA 1 system, available in portable form to engineers on the go. It shares the same high-quality spring steel and aluminium construction as the earlier NDH 20, but with an open-back design that helps to deliver a fast transient response and

maintains a natural, transparent sound throughout the entire frequency range. The linearity of the overall response is not achieved solely though the use of high-quality components: it also relies on the physical construction, which employs frequency-selective absorbers that Neumann say help to combat the overemphasis of high frequencies that occurs in many headphone designs. The attention to detail extends to the cable design, which has been optimised to minimise crosstalk between channels, and as with all Neumann products, the NDH 30 is manufactured to extremely tight tolerances to ensure a consistent performance between different units.

$ $649
W www.soundonsound.com/reviews/neumann-ndh-30
W www.neumann.com/en-en/products/headphones/ndh-30/

Sennheiser HD 800 S

Sennhseiser's HD 800 S design employs a large driver in order to maximise air displacement, which nevertheless has the rigidity required to deliver fast response times. Both the inner and outer edges of the transducer are secured to the drive unit, and the voice coil is wound in-house by Sennheiser to ensure that it is produced to the highest possible standard. The chassis and headband design not only provide a comfortable fit, but also incorporate some built-in damping and an absorber system intended to improve audio quality by tackling frequency masking issues. Each unit is manufactured to tight tolerances, and fitted with matched transducers that come supplied with their own unique frequency plots.

$ $1799
W sennheiser-hearing.com/en-UK/p/hd-800-s/

Shure SRH1840

Shure's premium offering comes in the form of the SRH1840, which feature a pair of drivers that are individually matched for each set to ensure consistency. The 1840 offers an extended high-frequency response, accurate bass reproduction and a wide stereo image, and is fitted with a steel driver frame with a vented centre pole piece designed to improve linearity and eliminate internal resonances. Comfort is ensured by lightweight construction and a dual-frame padded headband, along with replaceable velour earpads that contain a high-density, slow-recovery foam — a spare pair are included with each unit. Detachable cables are provided, and are securely held in place using MMCX (micro-miniature coaxial) connectors.

$ $499
W www.soundonsound.com/reviews/shure-srh1840
W www.shure.com/en-GB/products/headphones/srh1840

Sony MDR-MV1

The MDR-MV1 has been designed for mixing and mastering, both in traditional stereo and Sony's 360 Reality Audio spatial audio format — the latter can be achieved via the company's 360 Virtual Mixing Environment (360VME) software, and requires a visit to an approved studio to create a personal HRTF profile. It promises unparalleled spatial accuracy for all listening formats thanks to a precision-tuned open-back design loaded with drivers that have been optimised to deliver a natural, balanced sound with minimal distortion. Lightweight aluminium construction and soft, breathable earpads make sure it remains comfortable over long sessions, and it comes supplied with a detachable, replaceable cable.

$ $329
W www.soundonsound.com/reviews/sony-mdr-mv1
W pro.sony/en_GB/products/headphones/mdr-mv1

XCORE VALUES

Mark Lippett & XMOS

Most audio interface designs are based around technology from British innovators XMOS. What makes the xcore platform so ubiquitous, and what does it mean for musicians?

SAM INGLIS

Which manufacturer's products are found in more studios than any other? Whatever the popularity of Shure microphones, Apple computers or Behringer synths, the crown almost certainly belongs to XMOS.

Whether it's a portable laptop rig or a sophisticated multi-channel setup, nearly all musicians and engineers rely on USB audio interfaces. Yet although there's fierce competition between many well-established manufacturers, most designs are based around the same family of platforms. Lift the lid on a USB interface, and its beating heart will most likely be an xcore chipset from XMOS.

How did one company come to be so central to all of our music-making? CEO Mark Lippett fills in the back story. "We spun out of the University of Bristol in 2005 with a novel processor technology that was designed to deliver the sort of flexibility to software engineers that hardware engineers had become accustomed to in FPGA platforms. Our processor arrays can emulate all of the types of compute that you encounter in

Mark Lippett is CEO of XMOS.

embedded systems, including AI, DSP, I/O and control."

Hardware In Software

"The thing that really sets us apart is that the underlying processor architecture is fast enough and reliable enough to implement hardware in software," explains Mark. "For example, we can implement SPDIF, ADAT, I squared S: all of those protocols are actually software libraries to us, not distinct pieces of hardware on the chip. So, by deploying different software builds, you can effectively create different system-on-chip designs on an existing semiconductor platform, using software alone. Our objective was to give the embedded software community an efficient way of deploying software onto platforms in order to create fully integrated bespoke solutions with a rapid time to market.

"When you have a somewhat, dare I say disruptive technology, you're looking for market discontinuity — points of entry for the technology in the market. The one that we discovered very early on was USB audio. Apple were going to stop putting Firewire into MacBooks, and they said 'You're going to use USB audio from now on.' And the peripherals industry said, 'That's all very well, but there isn't a chipset for that.' Seeing the opportunity, we built a solution internally using our applications engineering resources. And the rest is history."

Bridge Building

The XMOS chipset performs the task that's most fundamental to any audio interface: it serves as a bridge between the various audio input and output streams, and the USB data bus. "Essentially, there's a collection of I/O protocols that need to talk to each other in a certain sort of segment — we're very good at joining them together. They might be at different sample rates and require some interim processing for one reason or another, but we can connect those things together. You are then able to select which combinations you want just using software. So, insofar as your PCB will allow you to do so, you could effectively do runtime changes to the I/O protocols that you're supporting."

This is the key advantage XMOS have over over rival technologies: the xcore chipset can easily be configured to cope with whatever I/O streams the interface designer wants to include, simply by loading the appropriate software onto it. If no such product was available, interface manufacturers would have to cobble together multiple hardware chips each dedicated to one individual function, such as sample-rate converters and ADAT transceivers, or employ field-programmable gate array (FPGA) chips. FPGAs are similarly versatile and are used by some high-profile manufacturers, but the barrier to entry is high as they are expensive and require specialist programming skills. By contrast, any software programmer with a knowledge of C or C++ can take advantage of XMOS's library code.

"It's a sort of chicken and egg situation," says Mark. "When the company was founded back in 2005, FPGA hardware platforms were becoming higher and higher performance and more and more expensive. They were chasing communications applications, and consequently, people in the embedded space and the consumer space couldn't afford them. And then there was no point in having an FPGA engineer on the staff, so FPGA engineers disappeared and now they can no longer program FPGAs. It was almost a self-fulfilling prophecy that FPGAs were not that accessible in that part of the industry. The other way of looking at it is there's probably 100 times more software programmers than hardware programmers. So if you want to make a very empowering creative platform available, make it available to the biggest community of creative engineers."

One Chip

"There are various different ways of interacting with XMOS technology," continues Mark. "You can take it as a processor and do programming 'on the metal'. Or we provide a library — an SDK, if you like — that's a whole stack of USB audio capabilities built around USB audio, and you can take the SDK and the tools and put things together in a Lego box style. You can deconstruct it and reconstruct it if you want, so you can switch on interfaces, you can change the number of interfaces and so on. It's quite a high level of abstraction. The majority of our customers use that, and that's where they get a lot of flexibility across their product portfolio by effectively rebuilding different configurations from the SDK.

"One of our internal mantras is to be the only chip in the box, and in many cases we achieve that. In some cases, for other reasons, there might be an application processor in there. If you've got a very sophisticated windowing display, or something that's clearly

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Drivers & Latency

From the user's point of view, software drivers are one of the most opaque elements of audio interface design. The confusion is partly one of terminology — strictly speaking, an ASIO 'driver' is more than just a driver — but in essence, most of us understand the need for low-level code that allows our audio software to talk to our audio interface. Generic driver code is built into the macOS and Windows operating systems, but it has its limitations. Core Audio on macOS offers acceptable low-latency performance, but only supports class-compliant USB interfaces and the AVB protocol. On Windows, meanwhile, it became standard practice in the late '90s and early 2000s to install third-party ASIO drivers to bypass the built-in audio protocols, as the latter did not offer adequate performance. Although that has changed and Windows now has integrated support for the UAC2 multi-channel USB2 class-compliant audio format, ASIO remains the *de facto* standard.

So, if you buy an interface designed to work with music recording software, you'll usually need to install an ASIO driver on Windows, and possibly an additional driver on macOS too. But further confusion arises because the driver is often installed alongside other software, most typically a control panel application that allows internal settings in the audio interface to be changed. These control panel applications are developed by individual manufacturers and often have a very different look and feel from each other. However, if your interface uses an XMOS chipset, as most do, the chances are that the driver code will actually be the same. You'll either be using the class-compliant drivers built into the operating system, or the ASIO driver developed by Thesycon to work with XMOS's chipsets.

"We don't actually develop drivers at XMOS," says Mark. "We partnered with Thesycon and they did the UAC2 drivers for us and they had an established reputation for building drivers for that space. Nowadays, UAC2 is supported by Windows, so the challenge isn't quite so great, but Thesycon did a great job of of bridging the gap for many years between people wanting multi-channel UAC2 and the arrival of support in Windows."

The widespread use of XMOS chipsets and generic drivers means that driver performance is perhaps no longer the yardstick it once was for choosing an audio interface. CPU overhead, and latency caused by input and output buffering, will be similar for all interfaces that use the same driver. However, there are other factors that can add latency, such as the implementation of any onboard digital mixer or routing matrix.

» going to lean on a lot of open source software, then generally speaking, you're going to want to be running Linux and running an application processor. But if it's a more basic user interface or a deeply embedded application, our ambition would be to be the only processor in there.

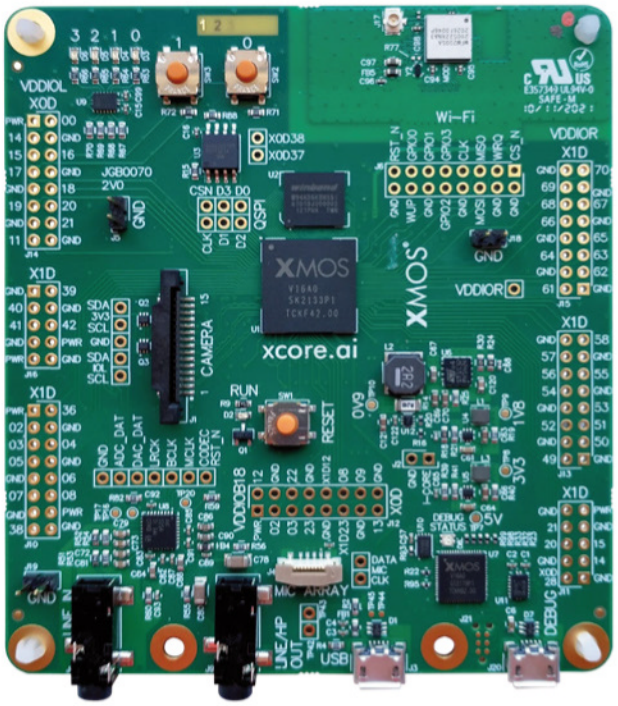
"There's a wide variety of things that people do with the processing that they can get their hands on, which is actually all of it. If you choose to, you can just peel everything away. I mean, Ableton's Push 2 has a fantastic display driver on it, which my understanding is driven by XMOS. And they did that. They're a great bunch of very talented engineers and that, I think, was a 'bare metal' implementation."

There have now been several generations of XMOS chipsets, and the company have recently announced a migration of their technology to the open-source RISC-V platform. "The reason for that is not because we've changed horses and decided to just build RISC-Vs. We've actually taken our existing architecture and made it RISC-V compatible. So we've still got this array of processors, but each processor is now essentially an extended RISC-V instruction set machine. It's still very unique in terms of the way it behaves and delivers very unique benefits."

Beyond USB

Although USB audio was the first commercial opportunity that XMOS exploited, their technology is equally applicable in other contexts. "We've also got customers building audio solutions in different markets. AI has come along in the last couple of years with tiny ML [*machine learning*] models that do things like keyword spotting, audio event detection, glass break detection, even gunshot detection. Those are audio or acoustic applications that are nothing to do with what we would traditionally have regarded as being an audio [*market*] segment. We've got all sorts of different applications for audio as a sensing technology, but actually interpreting it into metadata and using that metadata to enable other classes of applications.

"While there are a lot of legs in audio, you'll also see XMOS devices in applications that span the consumer, industrial and automotive markets, because fundamentally, if you buy a piece of silicon from us, it's completely uncommitted. There's nothing really on there that determines which application you move into. It just so happens that for commercial strategy, we selected a form factor and a cost point that fits neatly into the embedded audio sector. And actually the performance is appropriate for processing audio. While we undoubtedly have a great audio solution, it's not a dedicated audio processor.



You can't buy an xcore product off the shelf, but if you're an interface developer, you can work with an evaluation kit such as this.

"We tend to say 'USB audio' when we probably should just say 'audio', because in many cases our customers aren't using USB. Many of our voice technologies don't have USB. They're I squared S and I squared C on the back haul. Back in the day, we had the first standards-compliant Ethernet AVB interface. And that was an interesting one because we were in a head-to-head race with an FPGA company. We started three months later, and we beat them to getting the first compliant endpoint. And that just demonstrates the time-to-market advantage.

Stemming The Flow

One advantage of USB from the point of view of XMOS's technology is that the USB2 protocol has an inherently limited data bandwidth. Because this bandwidth is kown and can be accounted for, the xcore chipsets can be designed to cope with any possible USB2 data stream. That's not the case with more modern interface protocols such as USB3/4, PCIe and Thunderbolt, which are intended to permit massively fast, high-bandwidth data transfer. In these cases, an additional device is needed to filter out unwanted or unusable data and reduce the bandwidth to a level that the chipset can accept. "With Thunderbolt, we would need an external physical layer and, depending on what you're doing, the data rates might exceed our capabilities. We have had higher bandwidth interfaces, but we've always needed that external device to essentially 'drink from the fire hose' and just send the extracted data back to the xcore device. When you're into very high bandwidth serial interfaces, there's no way you can do it in a software pipeline in a processor. You need some dedicated hardware."

AI & DSP

On most interfaces, XMOS's xcore hardware doesn't just handle bridging. It also performs real-time audio processing, which is what the control panel applications supplied with your interface are controlling. The interface manufacturer is free to code their own signal-processing algorithms, but many make use of XMOS's own libraries. These include code that can handle audio mixing and routing as well as common processes such as compression, EQ and reverb. Most recently, the company have been focusing on integrating machine-learning tools, which has knock-on benefits for more conventional applications.

"With the third-generation xcore.ai, we put a vector processing unit into the architecture, primarily because we wanted to run edge AI models. But AI decomposes down to multiply-accumulate operations, with a couple of fancy things added on. So what we ended up adding was a very large SIMD pipeline for AI that also works really well with DSP. So now we've got a strong AI proposition, but also the opportunity to use those same

resources to do DSP. We've developed reverbs, compressors, you know, the sort of basic building blocks of some of these external sound cards. But now we've got much more horsepower and much more capability, more memory as well, which is important to start to really pull some of the more heavyweight DSP into the system.

"We're seeing a lot of customers are experimenting with converting DSP algorithms or DSP functions into AI. We're a very good platform for mixing and matching DSP and AI, because everything happens in the same place, so you don't have to export a load of data to an AI accelerator and then bring it all back again, reducing complexity, latency and power consumption.

"Our first sort of foray into that into that area was around voice processing: keyword detection, for example, on far-field voice processors. And we are also now seeing growth in opportunities in automotive, because if you look beyond the current generations to driverless vehicles, you're starting to see the cabin becoming an extension of a living room or office space, so again, there's more interest in high-quality, high-definition audio as well in those contexts. There's almost a new wave of audio applications happening now, a resurgence of something that we're all quite familiar with, but also some new use cases like automotive platforms and industrial defect detection. But people are also interested in AI in what I might regard as being a more traditional audio space, for signal conditioning and noise reduction and things like that, and we're a great platform for integrating that."

Moving Forward

XMOS's market dominance means that stable, versatile and affordable solutions are available to anyone who wants to build an audio interface, with no need to figure out the arcane lore of FPGA programming, code their own DSP algorithms, or integrate multiple chips to achieve the necessary functionality. But is there also a down side? Is there a risk that innovation is stifled when so many manufacturers are using the same platform? Mark doesn't see it that way. For him, XMOS solves one set of problems and in doing so allows manufacturers to focus on innovating elsewhere.

"The technology industry is so interdependent, and I think it's a question of picking where you want to innovate. XMOS may be doing all the USB audio bridging, and we're starting to bring the DSP in. But we're standing on the shoulders of giants as well. We've got tools companies that we use, we've got lots of open source technology that we use, we're using TSMC's silicon technology.

"There is a lot of innovation around bespoke DSP algorithms and bespoke AI algorithms, and we don't do those things in-house, but we're a great target platform. There's also the user interface that we're one or two levels of abstraction away from. There's significant creativity there. So I think it's just a question of picking your battles as far as technology is concerned and figuring out where you really want to differentiate yourself and partnering for the rest of it."

Sean Momberger & Nickie Jon Pabón: Jack Harlow 'Lovin On Me'

INSIDE TRACK

Jack Harlow's smash hit 'Lovin On Me' is the perfect marriage of old-school sample manipulation and 21st Century laptop production.

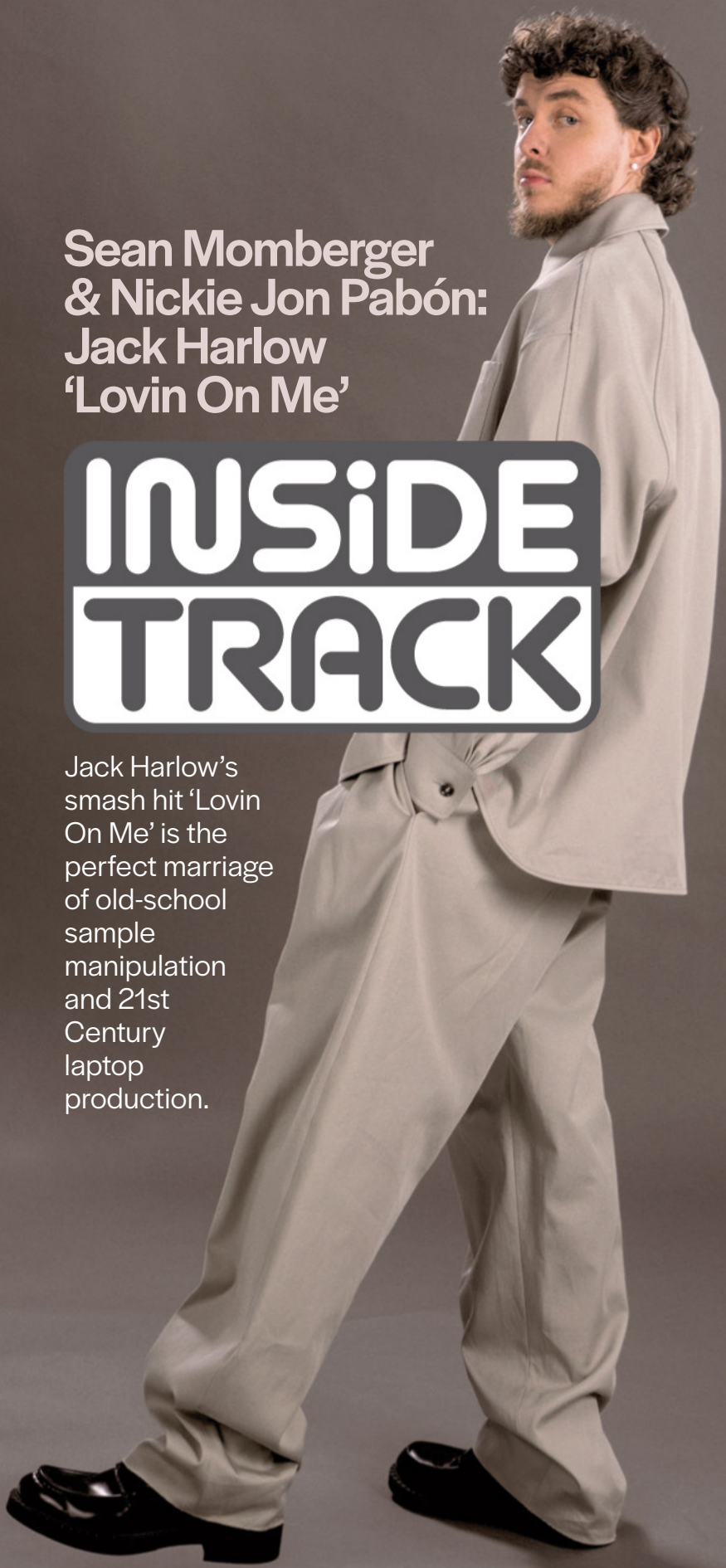


Photo: Cian Moore

PAUL TINGEN

Jack Harlow's 'Lovin On Me' topped charts in the US, the UK and many other countries. It was written and produced by Sean Momberger, with drums and 808 added by Ozan Yildirim (aka Oz) and Nik Frascona (aka Nik D). Nickie Jon Pabón recorded Harlow's vocals, added some writing touches, and mixed.

"I've been in the music industry for nearly nine years, and it's beyond my wildest dreams," enthuses Momberger. "This kept charging up the charts. It's been pretty surreal. Over the last nine years I've worked with big acts like Justin Bieber and Nicki Minaj, and people love it when you make a track, but when a track goes big like this, it's amazing how many new people you get to collaborate with."

Originally from Gainesville, Florida, Momberger moved to Los Angeles in 2014. When he first got into beatmaking, in 2003, he predominantly used hardware, but today he works with Ableton and mostly Pro Tools in home studio in a bedroom in his house in LA. He says that he makes 60 percent of his music all by himself, while the other 40 percent is made with others in the room.

"When I was in high school I Googled 'What gear do music industry people use?' and obviously Pro Tools came up, so I got an MBox, and worked with all the hardware. Then I got a bunch of VSTs, and made beats almost entirely in Pro Tools. A few years ago I also started using Ableton, mainly for programming drums, because Pro Tools does not have a good step sequencer. I prefer to start melodies in Pro Tools, because it's so open, it's like a blank canvas. I use Native Instruments Kontakt, and tons of Spectrasonics instruments, particularly Omnisphere.

"I run Pro Tools and Ableton on my MacBook. My MIDI controller is an Akai MPK Mini. It's cheap, but gets the job done. My soundcard is the [*UA*] Apollo Twin, which is amazing, and my monitors are the Yamaha HS8s. I have a Neumann TLM102 mic for when singers come in, and use a Sennheiser headphone for the vocalist, at which point I put on a set of Beats Pro, But I prefer to work on speakers. The MacBook speakers are so good, I sometimes make music just using them when I'm in a hotel room."

The Art Of Sampling

It was on this basic equipment that Momberger laid the foundations for what

The germ of 'Loving On Me' was a sample that Sean Momberger discovered and fitted to a beat and bass line.

was to become 'Lovin On Me', working entirely in Pro Tools. It began with a sample, which he says is a common way of working for him. "I'd say I'm 50/50 between starting with a sample, or with something I've written myself. In the latter case I just riff on my Akai MPK Mini, using different sounds and effects, and try to develop a melody that's pop. 'Hit My Line' [*a track from Chris Brown's 2022 album Breezy*] is an example of a song that I wrote like that. I just pluck around on the keyboard till I hear something I like. I simply go with the flow and with what inspires me.

"I also like starting with samples, because I am more into rap and hip-hop, and love bringing back old sounds. That's kind of how I made my mark. A lot of my music sounds not quite mainstream. Instead it has a little grit and nostalgia to it. This goes back to the late '70, '80s and even early '90s when everything in rap was sampled. I don't even think they cleared samples back then.

"When you take a well-known sample, it might catch people's ears faster. An example is when I sampled the Blondie track 'Heart Of Glass' for Nicki Minaj's song 'My Life' [*from Minaj's 2023 album Pink Friday 2*]. I made that with Don Cannon, and we sped it up, and Nicki killed that one. But I think people value it more when a sample is more obscure, because if it's not new to them, they just think you remade something. It's cool when you use an unknown sample, and people have to go back to listen to a song they never heard before."

Speed Merchant

In June 2023, Momberger says, "I found this old R&B song from Detroit from 1995, 'Whatever (Bass Soliloquy)' by Cadillac Dale. I loved the vocal phrase, 'I don't like no whips and chains.' It reminded me of a house/R&B type vocal. So I imported it into Pro Tools, chopped it up manually, looped it, and sped it up from 94 to 98 bpm, using Serato Pitch 'n Time. The sample became more of an earworm when I sped it up. It was super catchy.

"Speeding up a sample definitely makes it more fun and commercial. In general I prefer fast music over slow music so I tend to push the tempo. Also, in West Coast rap, bass lines are really important. Producers like Mike Mosley, back in the day, and DJ Mustard, have strong bass lines, and catchy vocal chops, and things are up-tempo.

"Then I put a bunch of Avid stock plug-ins on the sample. The sample has a bass line, but it's a little wonky, so I EQed all the low end out with the Avid AIR Kill plug-in, and then I pulled up an

Sean Momberger: "Speeding up a sample definitely makes it more fun and commercial."

Sean Momberger

"I'm from Gainesville, which is a small town in Florida, not much going on. I took drum lessons and piano lessons while in middle school, but I ended up quitting because I liked sports better. I remained interested in music, and was watching Kanye West, Just Blaze, Mannie Fresh, all these different producers online. It intrigued me, so I started making beats at home. I started out with an Alesis SR-16 [*drum machine*], and because I wanted to be like Kanye and saw he had the MPC, I got the smallest one that my mom would get me for Christmas, and started on that.

"Next I got a Fostex MR8 multitrack recorder that I would track my beats into. I also had a little cheap keyboard. After that I upgraded to my first sampler, a Korg ESX and got into sampling. I also got the Akai MPC1000, and that's when I really got into a whole new sound. Next I got the Roland, Fantom X6, which I still have. I was kind of using that when I got Pro Tools. So I was making beats on the Fantom and tracking them into Pro Tools.

"I went into music full-time, but for the first 10 years, I did not make any money, and one of my brothers believed in me, and helped me financially. My first break came in 2014. I had been making music with a producer who ended up working with Iggy Azalea. I was flown out to London, and they were mixing the song 'Fancy'. They were like 'Hey, you wanna add something?' I added five or six things, but only a small synth FX sound ended up on the record. I got some publishing, but was credited only as a keyboardist. The song became huge, and just having my name in the credits by adding a little synth sound really made a difference. A couple years later I got a big cheque from the royalties of the track and I was like 'Oh, wow.' Things went from there.

"I don't think managers and publishers will get you into any rooms that you can't already get into. They're going to help connect dots, but it's your track record that helps to get you into the right rooms. I was more of a slow burner. When I moved out to LA in 2014, I popped into sessions, you meet new people, this guy that knows this guy. After that it was a snowball effect."

» ARP Odyssey sound in Spectrasonics' Trillian plug-in and played in a similar but bouncier bass line. I also added some AIR Chorus, for more width and to brighten it up, the AIR Filter Gate and AIR Delay for some more bounce, and the AIR Reverb.

"The idea kind of wrote itself. In rap and hip-hop I think of a sample as a snake charmer. It's super hypnotising. In other genres you want different sections to come in, you want it growing, you want a beautiful composition. Rap is more loop-based. So if you have a super-powerful sample loop and tagline like the 'whips and chains', it's super compelling.

"I then exported what I had done, and sent it to Oz. A month later he and Nik added drums and an 808, and upped the tempo even more, I think to 105bpm. I had added some drums, but they were a little more open, closer to R&B. It wasn't until Oz and Nic put this up-tempo bounce on it that the track really popped. They get all the credit for making it fun and upbeat. Oz then sent the track to Jack Harlow. Jack went crazy over it, and told us not to send it to anyone else, and added his lyrics, which relate to the original sample. When I sent it to Oz, I wasn't thinking that Jack was going to get on this. It was just, 'This is a really cool idea, let me put it in our Dropbox folder, and let's see what he can do with it.'"

Infection Control

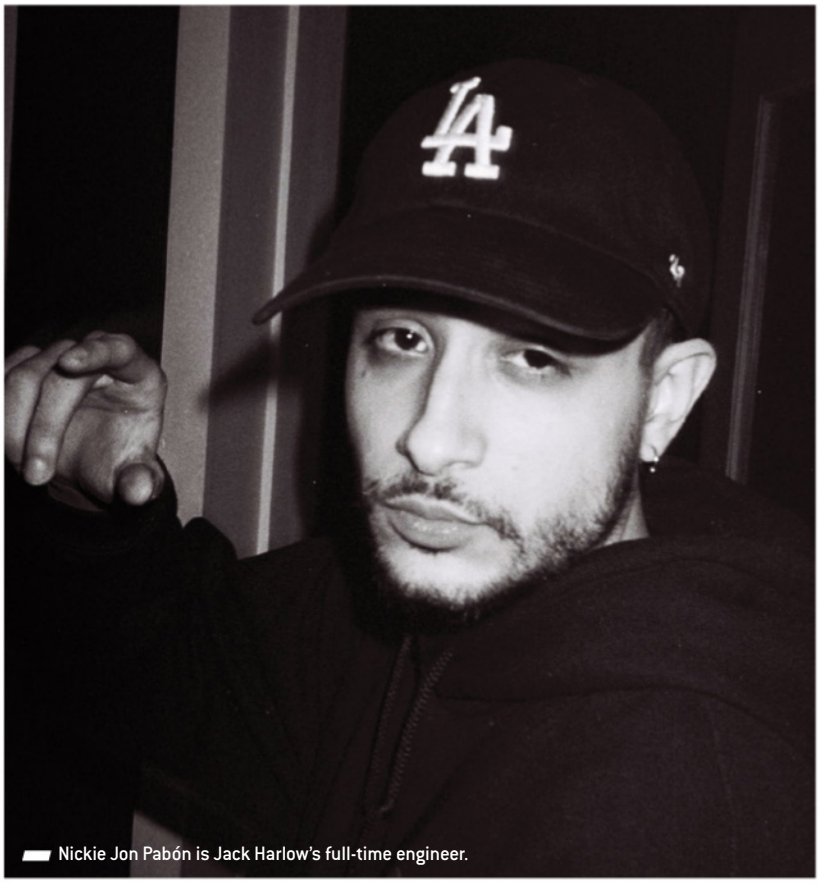
Jack Harlow's engineer and mixer Nickie Jon Pabón takes up the story. "The moment the beat came in, Jack fell in love with it. He immediately responded, saying, 'I'm hopping on this, don't send it to anyone else.' It was one of those beats that had the infectiousness that we were looking for at the time. We wanted something up-tempo and very catchy, and this beat met all the criteria in such a cool way.

"The sample sounds iconic, but it's actually not well-known. Even though it gives the feeling of being nostalgic, it's the first time most people will have heard it, and this made it a cool creative moment. Jack initially listened to it on his phone, and I think he instantly knew where he wanted to the vocal to sit, and this became the foundation of how we approached the track, both in terms of writing and mixing. You continually make an effort to enhance things, but we were always making sure that it remained connected to the original feel of when Jack first heard it."

Nickie Jon Pabón

"I was born in the US to two Puerto Rican parents. I've explored many different avenues with music and creativity throughout my life. My earliest idols were superstars like Justin Timberlake, Usher and Michael Jackson. As a teenager I picked up the guitar and I spent three or four years dedicated to everything that had to do with the guitar. As I went further, I dabbled with singing and performing, and acting.

"I got into the technical side seven years ago. In 2016, I decided that I wanted to go to SAE, the audio engineering school in Atlanta. After graduating, in 2017, I started working at Means Street Studios in Atlanta, Georgia, owned by Generation Now and Atlantic Records. I started as an intern, worked my way to engineering. I had been recording a little bit of everything there for a year, and then Jack got signed to Generation Now. By then, I was one of the main engineers, and when Jack got to the studio, we started working together, and we haven't stopped since."



Nickie Jon Pabón is Jack Harlow's full-time engineer.

According to Pabón, they had approached Harlow's previous worldwide number one, 'First Class', which is based on a sample of Fergie's well-known 2006 song 'Glamorous', with the same attitude. "Even in that case, we were pretty firm on being innovative sonically and not being too attached to the original. When you strip a piece of audio from something that people will instantly recognise, it's also about how you manipulate that sample. Once the beat comes to us and vocals are laid, it's up to us to create a sound world for it, and it might actually benefit us if it sounds completely different to the original. With both 'First Class' and 'Lovin On Me', the mission was to create something that hadn't been done before."

Go Anywhere

Nickie Jon Pabón has been Jack Harlow's engineer and mixer since the two met in 2018 at Means Street Studios in Atlanta. The two have been almost inseparable since, with Pabón travelling with the rapper while he is on tour, helping him with his stage performances and recording new tracks. Their collaboration yielded a Grammy nomination for Lil Nas X's song 'Industry Baby' (with Jack Harlow), and Pabón has also worked with the likes of Cardi B, Playboi Carti, Lil Durk, and The-Dream.

Pabón travels with the most minimal of setups. “I have a mobile studio that we take pretty much everywhere. It provides me with the continuity to work anywhere. We may open up a session in his hotel suite on the road to start a song, and then go back to Louisville, Kentucky, where Jack is from, and finish the song there. When we’re working and editing songs in different places, we don’t have to worry about it sounding different. We can work in big studios or we can work in hotel rooms.

“My mobile setup is always the same: my M1 laptop, Apollo Twin soundcard, and Audeze MM-500 headphones, which I mix on. Don Cannon lent us his Neumann TLM103 microphone, which sounds really good. I think Lil Uzi Vert was using it before us. We wanted a microphone that could withstand international travel, and while the Telefunken ELA M251 and Neumann U47 sound great, they are too cumbersome to carry around with you. I can just throw the TLM103 into my carry-on luggage.

“I also have an [*API*] Lunchbox, with a Shadow Hills pre, a Neve compressor and the Pultec EQP-1A, which is my recording chain. Plus I travel with two Audio Technica ATH-M50x headphones, one for each of us, for when I’m recording Jack. The Audeze are open-backed, so I don’t use them for recording. I also have a Little Labs headphone amp, which powers the headphones. The sound from the Little Labs is slightly different to when I plug my headphones straight into the Apollo, and I toggle back and forth, to refresh my palate, and as a reference, to see how the mix is translating. But I prefer using the Little Labs.

“I also carry the IK Multimedia iLoud Micro Monitors, because they allow Jack to listen to the beat in the room, and they’re not too loud, so we can record and not disturb the surroundings. We do a fair share of sessions in hotel rooms, and Jack’s management and I are on the same page in the sense that when they’re going to a book a hotel, we ask for pictures of their suites, and if I see carpets and drapes and not a lot of marble flooring, I know it’s somewhere we can work. Sometimes I have to put up some towels or blankets, but in general recording in hotel rooms is not much of a problem if you know what it looks like ahead of time.

> Nickie Jon Pabón: “The difference between various hotels rooms is easier to deal with than the difference between using one studio’s gear, and then going to another studio, with completely different gear, and trying to replicate the sound.”

“The difference between various hotels rooms is easier to deal with than the difference between using one studio’s gear, and then going to another studio, with completely different gear, and trying to replicate the sound. That can be a headache at times. So also when we are in a studio, we run everything through my mobile gear. If there’s an SSL board, I’ll have them route my system to a channel, and I’ll listen to the studio monitors for an additional reference. We tend to use studios when we’re working with producers and need more room.”

Hooks Up

When Harlow and Pabón received the beat for ‘Lovin On Me’ in July 2023, they were in a private recording studio in the countryside near Nashville. “We started the hook there and then finished the verses in Louisville. That hook came super quickly, we recorded it the day he got the beat, but the verses took more work. I think he wanted to really be precise with them, and it took a little bit more time for him and I to nail it.

“Every song Jack does is different, but in general he likes to write to the beat as it’s playing. He’ll rehearse it as he’s writing, so by the time he goes into the booth, many of his songs end up being first takes. We do edits on a case-by-case basis, but for the most part the main skeleton of the song is laid down in the first five minutes of him being in the booth. Then he comes out and listens to it, and will adjust things if necessary.

“There are times where he punches lines individually, and there are times where he might lay stuff down 10 to 15 times, and between him and I we pick the best takes between different words and phrases in different takes. I’ve always preferred the artist to do a bunch of takes that are just feel-based, and »

» then go into editing mode once there is enough material to work with, instead of trying to do a take so precise that you lose the performance and feel. When he sings, and he has a great singing voice, it's a different process and we like to experiment with stacks on his vocals, effects, etc. But for the most part, his songs are all first takes."

Mixing

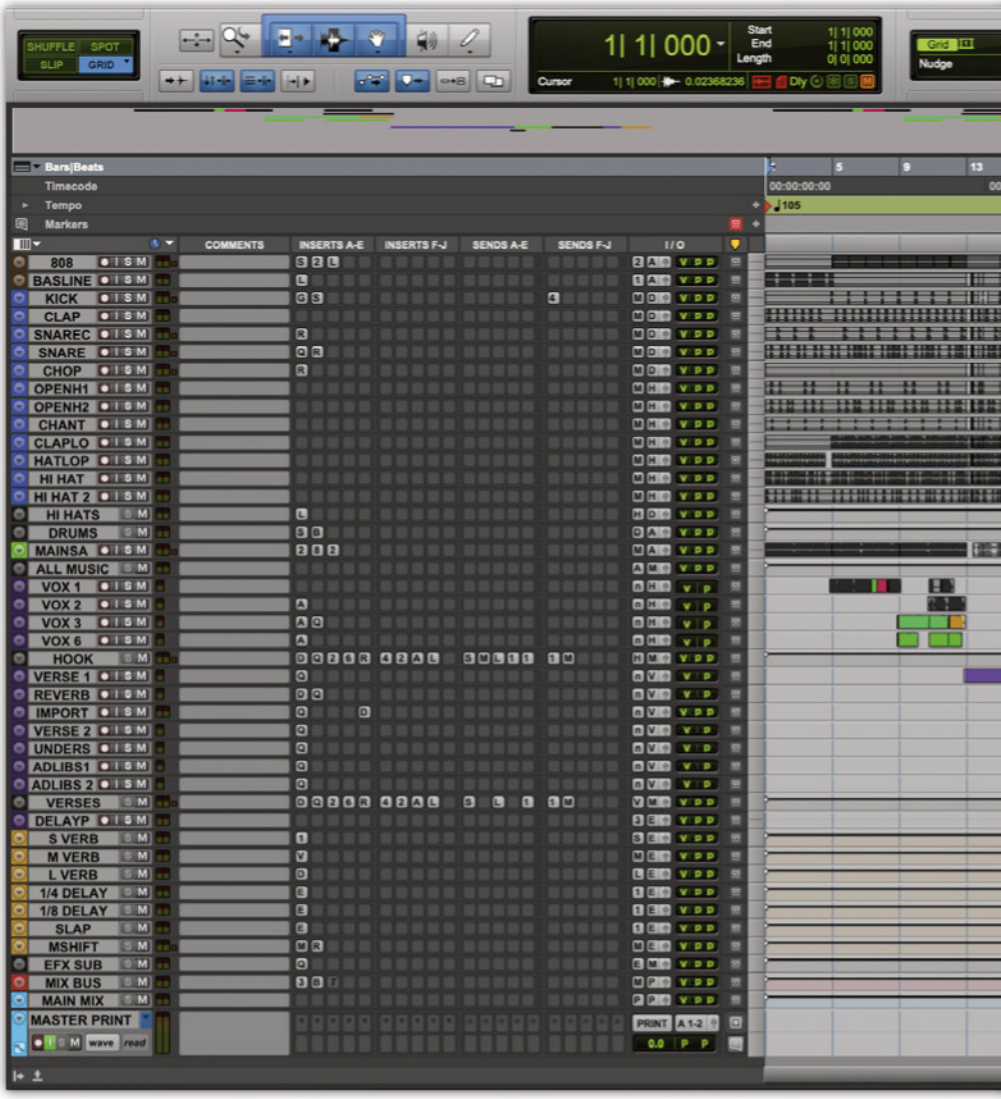
Rap music seems gradually to be moving away from specialist mix engineers, and 'Lovin On Me' is the first big hit by Harlow for which Pabón did the entire final mix. "Tracking and mixing run into each other in my process. When I'm in the studio with Jack, I want to give him the liberty to shape the song sonically exactly like he envisions it. That process happens throughout us working on the song. While writing the song on the first day I'm tweaking plug-ins as I go. Even when I'm using the same chain for a while, I still go in and play with the settings. The next day we'll have fresh ears, and then we may make some more changes. By the time it's ready to mix, he might still have feedback, and I'll incorporate that into what by then is a recording-mix hybrid session.

"I've always been involved with Jack's mixes, from early on. When Leslie Brathwaite or Jaycen Joshua were doing the final mixes, I bounced out vocal mixes. And Teezio [*Patrizio Pigliapoco*] and I mixed 'First Class' together, sitting next to each other. It's a good thing to be collaborative, and it has enhanced my skills. *Come Home The Kids Miss You* [*Harlow's second album, 2022*], was the first album on which I worked on mixes by myself.

"I prefer to start to start my mixes on the Audeze headphones, with the headphone amp, because that's where I get the bulk of my balance and creative ideas done. After that I can use any set of speakers, though I tend to take them with a grain of salt because I am constantly on the move and never get to used to a room enough to know what is real in the mix and what isn't. I do try to get other reference points, like the Audio-Technica headphones, the iPhone, AirPods, any studio monitors, car stereos and so on, but the Audeze is the most important to me for shaping detail."

Stealth Reverb

Pabón's mix session of 'Lovin On Me' consists of 15 tracks for the beat, 11 vocal tracks, several group aux tracks, seven aux effects tracks, an 'EFX' sub, a mix bus and a main mix track. There are just a handful of plug-ins on the beat tracks, but the hook and verse aux tracks are heavily loaded.

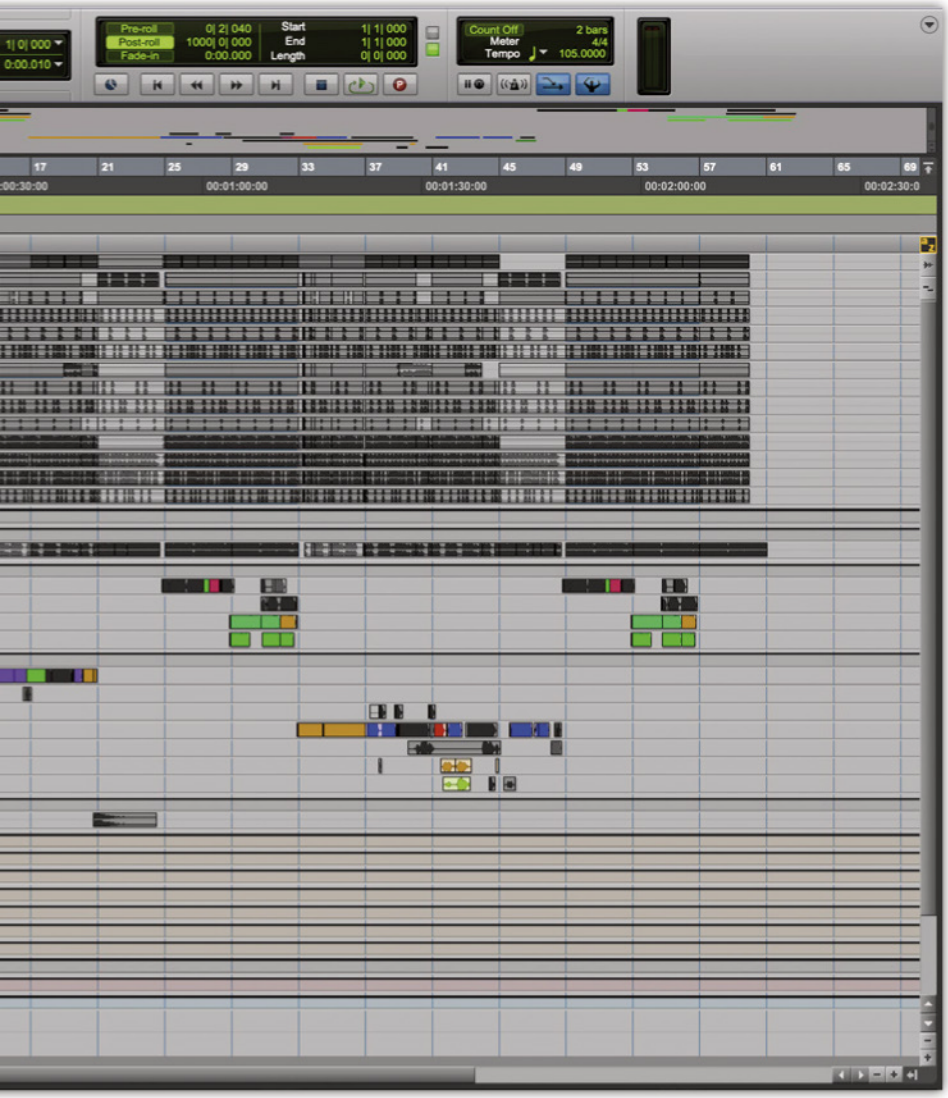


By modern standards, the final Pro Tools session for 'Lovin On Me' is unusually compact.

"We made a couple of arrangement moves with the instruments. Jack gives his feedback on what he wants out of the beat as he's creating, and in this case there's an edit in the last four bars of each verse, in which we muted parts in Pro Tools. The plug-ins on the beat are me putting some last touches on the sample, just light polishing. I enhanced the 808 and the bass with the Little Labs Voice Of God for more definition. I also beefed up the kick with Mike Dean's Gain Station plug-in, and Transient Designer, to get more separation. The 808 is constantly there and the kick comes in at the top of every 'one'. So I gave the kick some more character and definition.

"I wanted to get a little bit more out of the sample from what Sean had given us, using the SSL Channel Strip 2 plug-in. I boosted some high end to give it a bit more clarity and definition, and then I ran it through an UAD Studer 800 [*tape emulator*]. The last plug-in is the Oeksound Soothe 2, lightly taking care of any harsh frequencies that may have come up with the saturation from the 800.

"The first five or six plug-ins on the hook and verses aux tracks are from my recording template. Those are plug-ins that I use while I'm recording him, so it's the sound that he gets used to while writing and recording. In this case, the first one was the Waves De-esser, then the FabFilter Pro-Q3 is doing surgical EQ, followed by the SSL Channel Strip 2, the Waves PuigChild 660, and the



Waves RVox. One plug-in I added during the session was the McDSP MC404, for multiband dynamics.

"The next plug-ins are final tone shapers: the Soothe 2, the Eiosis AirEQ for some more air, and the last one is a limiter called Limitless by DMG Audio, which is for a final push to get the vocals to cut through. These final three plug-in are my mix plug-ins, that I would not normally use for recording.

"The aux effect tracks are also part of my template, and the settings get changed for each. They serve different purposes. The small reverb comes from the Softube TSAR-1, and is a very tight reverb that gives barely the perception of a room. I like using it in rap, because when recording vocals that are so punchy, it helps me fill in the space in between words without taking up too much actual space in the mix.

"The medium reverb is from the Valhalla Vintage Verb, and is a little bit more audible, and the long reverb is the Avid D-Verb. I also love using the [*Liquidsonics*] Seventh Heaven plug-in for long reverb. You can't really hear a lot of reverb on this song, so it's really just how the reverbs are accentuating certain tones within his voice. It's almost like I'm doing EQ with the reverbs. Jack's vocals don't have an atmospheric sound, so most people don't believe that I have three reverbs on him!"

'Lovin On Me' was released on November 10. A week later it debuted at number one in the UK, and another week later it reached the top in the US. Harlow called the song "a new era" in an Instagram post. Its enormous success is also likely to herald a new era in the careers of Momberger and Pabón.

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TalkBack

Noema Te Hau III

WILLIAM STOKES

Kiwi producer and songwriter Noema Te Hau III is based at Big Fan Studios in Morningside, Auckland. The groundbreaking not-for-profit, multi-purpose facility was founded by Joel Little, who has worked with some of the biggest pop artists on the planet, including Taylor Swift, Niall Horan, and fellow Kiwi Lorde. Noema was educated at MAINZ, the Christchurch music and audio institute where Little also trained. Not long after graduating, he was soon back at the Institute, this time lecturing in production, music theory and performance. It was then that Noema got the call to help Little set up Big Fan.

At the moment I can't stop listening to

At the moment I can't stop listening to a song by Unknown Mortal Orchestra, who is a Kiwi artist but based in LA. I'm weirdly such a big fan of Kiwi music at the moment — it never used to be that way. Maybe it's because I'm a bit more involved in the New Zealand music industry now. But yeah, Unknown Mortal Orchestra, the *V* album. I love 'Layla'. It just feels good. It kind of gives me Fleetwood Mac vibes. It's pop, it's digestible, but he somehow bends things like harmony a lot. The way he plays guitar. We do a lot of writing camps here in New Zealand, and I know Ruban [*Nielson, UMO*] has done a couple but I haven't run into him yet. I'm sure it's coming!

The project I'm most proud of

In terms of music, I just finished an album with an artist called Alayna. She's another Kiwi artist. We studied together at MAINZ and our careers have kind of kind of grown parallel to each other, her as an artist and me as a producer. It was a debut album, and we had a decent budget to get it done. It's very much a concept album, which I'm a big fan of. So it was a grind! I think it was about three years, or something like that. One of those things where we just left no stone unturned. So yeah, I'm very proud of that.

Apart from music itself, it would be Big Fan. Managing the studio here. I got to help, you know, start this place. I got to set up all the studios, I got to go through all the little things that you forget are at the start of a new venture. So I'm very proud of what we've done here. It's a charity. We have a two-storey building here in Auckland, in Morningside, so it's pretty central to the city. The bottom floor is a live venue, which has a capacity of 170, but it's a very good music venue. It has an in-ear monitoring system, a really high-end PA... Essentially, Joel's vision was that big artists could come and do a really small show here, and then high-school bands could also come and play, using the same stuff that those other bands have been using, more or less.

And then the studios: we have three upstairs, and there's a fourth one which is Joel's studio. He works out of here a lot as well. They're basically production suites, so they're not huge, but they're very well-treated rooms and they have everything that you would need nowadays: some preamps, maybe a compressor. For example, this studio I'm in has some Chandler TG2s and some API pres, all just running through a [*UA*] Apollo. We have some Focal speakers. This room has a drum booth. It's based around creating music, essentially. For writing, for production. All the rooms have their own characteristics. But they're all based on what Joel uses, just a simple setup.

The first thing I look for in a studio

When I'm going into a studio it's usually to create, rather than just engineer. So I'm looking for a setup that's comfortable and creative: ideally, one that's already set up to be as creative as possible. So, I'd like a drum kit to be there, already miked and ready to go, a piano there, ready to go, a vocal mic to be set up... I quite like having things around me. So sometimes, I won't even set up in the control room. I'll set up in the live room, kind of in the middle of it. I'm very much there to create, first and foremost. I quite like that. Maybe we'll set up a couple of couches in the live room — I've done that a few times for sessions. That's quite fun.

If we're talking about the fundamentals, building a studio, the first thing I'm thinking about is treatment. When I was building this studio I had never done that before, I'd never done the testing with a proper company before. Seeing these rooms empty, and literally just bringing treatment in and throwing it on the ground — it was night and day. Before even placing it! It was like, holy shit! If there's anywhere you should spend money first, it's there.

The person I would consider my mentor

To be honest, I never really had a mentor at the start of my production career. I was always kind of annoyed at that! I was like, "God, damn. I just wish I had a bit of an in, to be able to learn off people!" It was always very self-driven. But now, it would definitely be Joel. He shows us some of the Taylor Swift stuff, some of the big stuff he's working on. He'll show us the sessions, break them down with us. He'll show us how he'll go into sessions as a producer, how he prepares. He's by far my biggest mentor now. And he's also just a bro! So it's easy to chat to him. He's also brought in other engineers and producers, his friends, for us. So, for example, Mark Rankin is one of his really good friends. A crazy good engineer. He did a lot of the Queens Of The Stone Age stuff, some of the Adele stuff. I think he did a lot of the *21* album. He came and stayed in New Zealand with Joel. One day Joel was like, "Man, we all suck at recording drums! We should get Mark in!" So Mark spent a day with us, just showing how he would do drums in our spaces, with the gear we had. He worked his overheads first, with the kick and snare lined up so they were dead in the middle of the image, then used everything else to supplement — a few tom

mics if the song needed more toms... It was really cool. I loved it.

My go-to reference track or album
I can't say that I necessarily have one go-to reference track. When producing, I mostly let the artists I'm working with lead with their own reference tracks! What they're currently listening to, or music that they love. Then I'll do a bit of listening analysis to break down the arrangement, tones and production techniques being used. But if I really had to pick, it would either be 'Teenage Dream' by Katy Perry or 'Human Nature' by Michael Jackson.

My top tip for a successful session
It's reading the room, essentially. Especially if you haven't worked with someone before. You have to learn, really fast, what kind of person they are. Some people like to talk first. Some people just like to get in and smash stuff out! So yeah, you really have to be able to read the room and be flexible in any situation. If you can research the person you're working with beforehand, do that. If you know other producers they've worked with, maybe just ask them a question, or something. Being prepared for anything that comes your way. Sometimes that looks like a template: for me, if it's a writing session, I've got a writing template that I like to use, just to make things quicker. Just be prepared for anything!

The studio session I wish I'd witnessed
I'm such a big fan of 'Toxic' by Britney Spears. The arrangement in that is just so weirdly eclectic, like, it has these like spaghetti western guitars, these crazy string lines, and I just want to know how on earth they thought that would all work together! I'm just obsessed with that arrangement and how strange it is. But also how fucking perfect it is at the same time. And of course you don't notice it because your attention is on the vocals, the whole way through. But I'd like to have been there when they were putting all those ideas together.

The producer I'd most like to work with
That's a tough one. Max Martin has been someone I've studied so much. But then again, I'd also be so scared to work with him! I guess I idolise Max the most. But it would probably be someone like Dr. Dre, or Pharrell. I just think they have such an interesting approach to their arrangements. I like the grooves they create. Especially Pharrell. It's weird: I don't make a lot of hip-hop stuff. But I think that'd be like a really fun experience. I think I'd learn a lot working with someone like that.

The part of music creation I enjoy the most
I was actually talking to Joel about this recently, and we both kind of agreed that it's in that initial songwriting, early production phase, where something just kind of hits. The song just finds its home. It's that weird little bit of magic where you don't quite know what just happened. Suddenly, you guys are all in sync, in the room together. You just link up on an idea. And the idea just happens to be amazing. It's the feeling I'm constantly chasing, I think, in every writing session. When you're like, "Oh, my God, we just came up with the most perfect idea for what we were trying to achieve." It could be a lyric, it could be a melody. It could be a riff, it can be lots of things. But sometimes everyone just lights up, like, "Shit, that's that thing that brings the song together!" Then you just feel like you don't want to get in the way.

The advice I'd give myself of 10 years ago
Firstly, trust your taste. In general, what I like about most artists, or most producers, is their own specific tastes. And that was something I never really got, early on. I was always trying to be quite technical, and things like that. But I now realise taste is everything. And everything kind of supplements your taste, supports your tastes. Like, you get better at working synths to showcase your taste. You get better at picking out samples to showcase your taste. So, yeah, I feel like taste is everything. And after that, it's just to keep working. Keep working hard. You have no control over what comes in your career. You really don't. So just do music because you love making music. I wish I had realised that earlier on. Whatever happens, happens. You have no control over that. Especially when you're a producer: so much is out of your hands if you're an artist, but if you're a producer, it's up to the artists you're working with as well.

How I Got THAT SOUND

Doug Showalter ▪ Harry Styles 'As It Was'

JOE MATERA

American music producer, songwriter and multi-instrumentalist Doug Showalter has worked with a wide range of artists across multiple genres, from Harry Styles to 30 Seconds to Mars, Smokey Robinson and Gabriel Black. Here Showalter breaks down how he crafted the guitar sound on Harry Styles' mega-hit 'As It Was' in his Nashville studio, Mt. Molehill.

"My approach to recording guitars has evolved a lot over the years. When producer/songwriter Tyler Johnson asked me to contribute some guitars to *Harry's House* I applied my current approach, which is to simply hit Record, jam along with the song a few times and improvise whatever ideas come to me with a few different guitars and effects.

"This style of capturing guitar parts, I believe, comes from my obsession with sampling. My love of sound runs tandem with my dedication as an instrumentalist, and over the last 10 years I've gotten heavily into sampling vinyl records, drum machines and cassettes as well as capturing sounds with my phone. Up until getting into sampling, my style of recording guitars was very traditional in making sure I always had my parts very well-rehearsed before hitting record. All that time spent sampling really informed an approach to recording guitars that ended up working well for 'As It Was'."

Sample & Hold

"The guitar that ended up making the record was a G&L Legacy Strat through '78 Fender Vibrochamp with a Strymon El Capistan delay pedal. On the engineering side, I used an AEA N22 ribbon mic running into a Shadow Hills Mono Gama microphone preamp. I feel the tone of the Strat mixed with the vintage tape sound of the delay pedal really complemented Harry's vocals during the second verse of the song.

"For the second verse of the song, I captured two guitar parts including a sliding octave melody and a scratchy, more rhythmic part. Both ideas I only played once while improvising. Those phrases were then used to create samples that I looped throughout the second verse. I pitched up one of the octaves samples using Pro Tools to match the chords of the song. This helped create a melodic phrase that sat nicely in the arrangement.

"I always track direct and miked signals when recording electric guitar parts; this was especially helpful working on this particular song. Once I had the parts I liked, I ran the direct signal of both guitar parts through a Strymon Big Sky reverb pedal back into Pro Tools. This helped achieve a whole new texture for all the guitar parts when I combined them together. The end result was four total tracks creating one overall sound.

"All the sampling, re-amping and use of effects helped achieve what I believe is a very unique sound. From the moment Tyler heard the guitar parts, he was super excited to get them into the track. I couldn't be any prouder of my contribution to this song."

Hear The Sound

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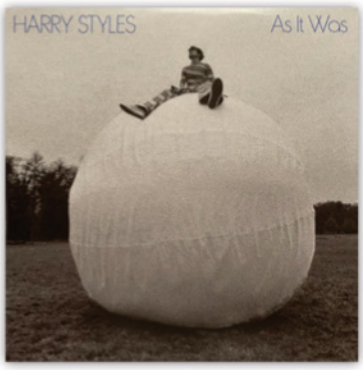
W https://open.spotify.com/track/4LRPiXqCikLIN15c3ylmP7



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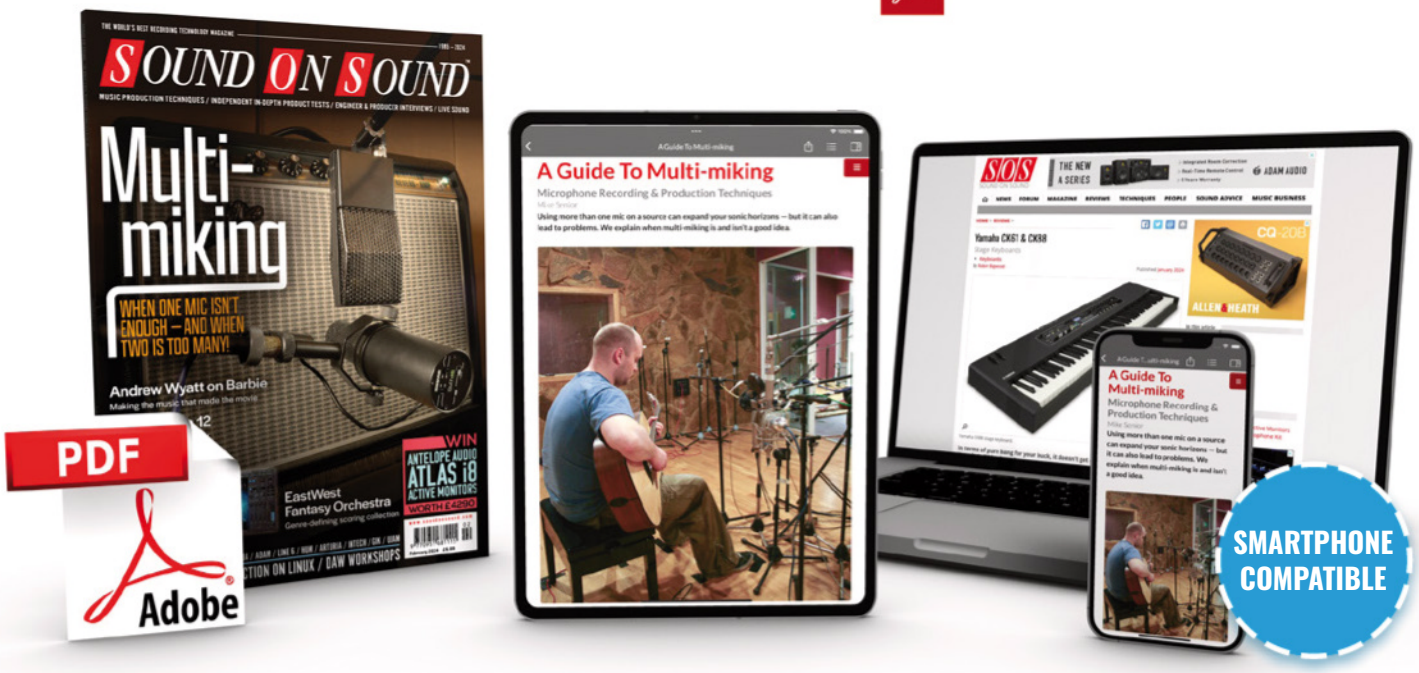

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BLACK ARTS

Producing Norwegian Black Metal, Part 2: Kark & Necromorbus

Eirik 'Pytten' Hundvin's work with Mayhem continues to inspire producers, 30 years on. Two of the genre's leading lights explain how they are taking black metal forwards.

Producer and musician Kark leads a new generation of torchbearers for Norwegian black metal.

Photo: Nicolai Karlsen

Kark Studios is big on atmosphere!

Photo: Nicolai Karlsen

JILLIAN DRACHMAN

"Black metal is art in its highest mode of expression. To me, it has always been synonymous with total musical, spiritual, creative and emotional freedom." Kark is best known as the guitarist, bassist and vocalist of Dødsengel, the Norwegian band that represents his shared vision with lyricist, drummer and scholar Malach Adonai. Dødsengel have greatly expanded the genre with their radically individualistic art, which transports listeners to new realms through the use of unusual instrumentation, unexpected ingredients such as components of classical music, haunting and often cinematic atmospheres, and mind-bending, acrobatic versatility. Dødsengel's second album, the otherworldly *Mirium Occultum*, is considered one of the best black metal albums of all time.

Dødsengel's *Mirium Occultum* is a Norwegian black metal classic.

Kark handles all aspects of Dødsengel's production, and is also an in-demand engineer and producer, working mainly with underground acts. His clients have included the legendary Manes, Behexen, Djevelkult, Askeregn, Jared Ambience Inc (the solo project of Seigmen's Sverre Økshoff), record label Terratur Possessions and countless others. Located in Ikornnes, Sykkylven, Kark Studios offers mixing, mastering, re-amping, recording, audio restoration, proximity to the mountains and eloquent company. At present, Kark is constructing a second building with an even bigger room for live performances. The addition "will have the same aesthetics as the existing one with a heavy focus on atmosphere. I have always felt that recording studios should be a place for creativity and recreation, a place that feels like home." Kark Studios' decor conjures dark romantic vibes and a dreamlike visual experience.

Raw Power

The work of Eirik 'Pytten' Hundvin, profiled in last month's *SOS*, has been a constant influence on Kark. "Pytten is essential. In a way, you could say that my approach to working with black metal recordings, and even other types of music, is built upon the legacy of Pytten's work. I suppose that in most art, in a sense, you have to 'stand on the shoulders of giants' to reach a new level beyond the old masters. You have to keep one foot in tradition and another foot in innovation. Thus, I take the old-school way of recording and work that into a modern setting." Kark has learned from pioneers like Pytten that "the production is as much a part of the music as the music itself. In most other types of music, I feel that people mostly speak of sound in terms of only good or bad, and that everything is usually oriented around being as hi-fi as possible."

Kark believes "the idea that you have to choose between having a full-range punchy sound and a more lo-fi sound" is a total misunderstanding. Rather, both options "are completely compatible with each other, and should be embraced instead of shunned".

As a musician and engineer, Kark strives to "combine technique and emotion so that they work together to bring out the full potential of each piece. Rawness is one of the key elements of black metal. Yet the kind of rawness is not limited to just one type of expression — going full-on mono and cultivating a sound that almost could have come from a faulty tape recorder. All of Dødsengel's recordings, for instance, have varying styles and degrees of this element. This can be summed up by having the sonic aspect match the emotional rawness. An example of this could be that very passionate vocals can drive a tube preamp to the breaking point, and this element becomes a part of the performance, rather than seen as something 'wrong'. Another example is not to polish the guitar sound in a manner that removes the ferocity of a high-gain amplifier. Rawness should not only be found in the actual performance, but also in how the performance is captured."

Going Bad

As one might expect, Kark's decisions regarding equipment depend on "taste, atmosphere, and the chosen colour. It's not about 'technical quality'; it's about what fits and what you want." He notes that although Pytten has always operated out of excellent studio spaces, he also made strategic use of "very bad gear, which gave a very unique sound. That concept can be turned in any way desired to achieve some very interesting results. So, when I record guitar, I use my Peavey Envoy 110 amp, which I have had since I was a kid. Then, I combine it with something like a Mesa Boogie MkIV and blend the tones to get something unique. So, you have the very dirty side of it and the very sophisticated side of it, and together they make something very special.

"This setup is something of a standard in my productions, regardless of whether or not they are my own, or if I am working with someone else. The usual setup is to have one of the amps miked with two SM57 microphones, using a Fredman mic clip. This clip gives a combination of on-axis plus off-axis mic placement, which gives endless possibilities in the blending of light and dark in the guitar sound. This is usually for the amp with the most gain. In combination with that, I split the guitar signal with a simple splitter box, and the other signal can go into another amp with less gain, which will be blended to complement the more high-gain amp. This amp will usually not be miked up but will go into a Palmer PDI 03 speaker simulator. I am not that much into using pedals, and prefer the amp's distortion. »

Kark's Mesa Boogie combo is often paired with a lowly Peavey practice amp and Palmer speaker simulator to create his distinctive guitar sounds.

Photo: Nicolai Karlsen

» However, I can sometimes put something like a Tubescreamer in front of the amp, or even use another distortion pedal, and utilise only its tone colour, not the actual distortion section.

"For the bass guitar, I split the signal and use a Mesa Boogie Subway DI and whatever high-gain dirt amp I see fit. Usually, an old Peavey Bandit does the trick. It is key to have the bass come through as both distinct in its own frequency range, yet somehow gel completely with the low end of the guitar. This is achieved by a blend of this description.

"Vocals are recorded through a Universal Audio LA-610 MkII with a medium peak reduction rate, and smoothed even further during the mixing with a Hairball Audio 1176 Rev A compressor. The mic is a [*Shure*] SM7B."

"Drums are the foundation, and their weight determines the weight of the rest of the instruments that go on top. If the drums don't have enough of a full frequency range, then that will give very direct limitations to every other instrument as well. In other words, the drum sound, to me, dictates every other sound that goes on top. I like to create depth in the soundscape by using different preamps for the different 'groups' of the drums, so that while the general sonic aspect of the drums has an evenness to it, I also introduce variation within the evenness. For example, I can use my custom API console for the kick and snare drum, while the toms go

Across The Border

Tore 'Necromorbus' Stjerna operates from his own Necromorbus Studio, also referred to as NBS Studio. Founded in 1995, it's situated in Söderfors, Sweden. Stjerna started out using a simple four-track, and built his first studio in his childhood home with the assistance of his father. The fourth and current incarnation of NBS Studio formerly served as a Pentecostal Baptist church. Although Stjerna acquired this beautiful building years ago, Covid-19 complicated his move. The renowned Swedish acoustician Ingemar Ohlsson has been helping Stjerna realise his vision. Stjerna tells us: "I love this new studio space. It's really the way I always wanted to have things. After basically an entire career of working in less than optimal rooms, I wanted to build something without compromises. I had heard about Ingemar before, but, for whatever reason, I thought his fee would be too high, I guess, considering his fame as a designer and his previous work. After a couple of failed attempts with other designers, I decided to give Ingemar a call. Now, I like to think that I am usually very straight to the point in conducting business, but Ingemar definitely beats me in that regard, so, for me, the choice was obvious from that point. I described what I wanted, and when I received the first draft, it was basically 90 percent there already.

"You never quite know what a room will sound like until it's finished, but when we finished the control room and fired up the speakers, everything was just perfect. Ingemar came over to measure and got great readings. I haven't had to change one single detail! The ceiling height in the hall is around seven metres, and I've kept it very open in terms of sound treatment. There's still plenty to be done there, but I've done several recordings there already, and I love the sound. I have room mics placed and mounted in various places around the hall, so you can play around and balance those in a way that I've never been able to do previously. I have a fairly large iso booth if you want a more controlled ambience. There's one more smaller booth in the design drawing, but I haven't got to that yet. Lastly, the area where the altar used to be can be closed off to some degree for extra separation, and it has a bit more treatment than the rest of the hall. So, there's a nice palette of options.

"I abandoned the idea of large mixing desks many years ago for various reasons, but I've been through several different controllers over the years to maintain a hands-on feel to finally land with the Yamaha Nuage setup that I'm using today. Cubase is my main DAW, so the choice was pretty obvious, I guess. I'm using Genelec 1038 speakers as mains, and the control room was also designed specifically for those. The main front end is an assortment of CAPI preamps and a few Neve clones from Sound Skulptor. I've been getting more and more into electronics over the years, so I've built a lot of the equipment I have here today. One of the latest additions is a bunch of SSL EQ clones from Link Audio Design. Various other studio staples, of course: some 1176 compressors, a couple of [*Empirical Labs*] Distressors, a couple of Pultec-style EQs, and so on. I've also built up a bit of a collection of backline over the years. A bunch of nice drum kits and snare drums (I'm a drummer, originally, so I guess it comes naturally), a lot of guitar amps... I have way too many toys, but, luckily, I have a bit of space for it at least."

through an old Peavey 701R mixer, and the overheads go to an old Behringer Eurorack mixer — a blend of sonics that make up a unique whole, brings forth the nuances, and blends gold with grit for a dirty golden flavour. As for mixing drums, I use a lot of parallel processing in order to maintain the original signal. Yet, I blend the flavours to taste, using two dbx 160A compressors. For drums, a crush bus [*aggressive parallel compression*] is something that is close to being a must."

Free Your Mind

When it comes to mixing, Kark says that liberating himself from preconceptions has been essential to his development. "Rules that have been important to break include how much EQ you are 'allowed' to add to a certain element. If the kick needs a 20dB boost at 60Hz, then I will do that. I listen to the mix as a whole, and I will not solo each and every track to remove things like 'boxiness' and so forth just for the sake of it. Some of these elements that are generally considered 'wrong' could actually be a big part of what makes a mix

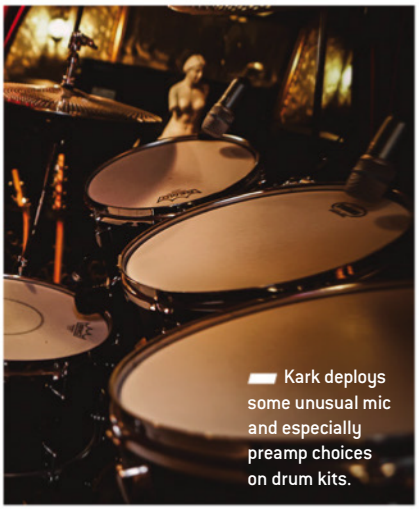

Photo: Nicolai Karlsen

Kark deploys some unusual mic and especially preamp choices on drum kits.

sound unique. A mix is a combination of sounds and the sum of how they work together. A comparison could be like this: The paintings that are the most alive are the ones where the lines are slightly blurred, and they have a slightly rough texture. This can be found in the paintings of Odd Nerdrum, for instance. The same goes for a mix. There should be a clear definition and separation of elements. Yet, at the same time, they should melt into each other to create a cohesive whole.

"The mixing starts at the tracking stage, and the mastering starts at the mixing stage. In other words, there is always a certain idea of how everything will end up, with as little as possible left to be sorted out later on or 'in the mix', as they say." Kark aims to create "music that actually opens up when you turn up the volume, instead of just causing you ear fatigue. A good sign of a mix and master done the right way is when you find yourself turning it up more and more as you listen to it. Mixing and mastering at more conservative levels, as the loudness wars are over, does really make way for all the details to sit right in the mix when you crank your stereo."

Kark's passion for engineering has always been bound up with "the magic of being able to preserve a moment in time". While Kark was playing in his first band, he considered recording in a nearby studio. However, he and his friends were warned that its operator would try to restructure their songs. "So, I bought my own

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Tore 'Necromorbus' Stjerna is a leading black metal producer and engineer who operates out of his own studio in Söderfors, Sweden.

» equipment and started my own recordings to have control."

He acknowledges that producing his own material can create a challenge due to a lack of distance. Nevertheless, it has ultimately helped him as a performer: "When I'm recording guitars, bass and vocals, I'm alone in the studio. I don't have a second set of ears to rely on. So, it's an exercise in decisiveness, and it's an exercise in trusting yourself."

Necromorbus

Tore 'Necromorbus' Stjerna has earned a stellar reputation as one of black metal's leading studio and front-of-house engineers, an excellent musician, a highly sought-after manager, a former record label owner, and more. Stjerna has worked with hundreds of clients in the extreme metal world and beyond, including Shining, Behexen, Deströyer 666, Triumphator, Desultory, Jess and the Ancient Ones, Portrait, Tribulation, Unanimated, Malign, Valkyrja, Inferno, Voodus and so on. He has recorded several of the bands with whom he has played, such as Ofermod and Funeral Mist, and has performed live with metal powerhouse Watain, having acted as their producer since 1999. Stjerna contributed session drums to Armagedda's *Ond Spiritism* between producing their classic second album, *Only True Believers*, and remastering the equally memorable *The Final War Approaching*.

"I'm looking very much in the rear view mirror when it comes to black metal, at least," says Necromorbus. "I'm very much in the '90s." It's a reference to the "distinctly different paths that black metal went down, with the extremely DIY approach and the more polished alternative. The idea was to do everything in a very primitive way, but it was still recorded in a 'proper studio' and mixed by a legend. I think that marriage between the two approaches is important. I focus on trying to get the band's identity to really shine through, and having a good variation."

Overdoing Perfection

This philosophy stands in sharp contrast to much modern metal, which often has a homogeneous and over-produced sound. "We have somehow got to a point where everybody seems to be walking down the same road. It's not that everything sounds the same, but there's very little variation. Death metal exploded in my formative years, and there was an insane amount of creative freedom back then in terms of both the music and the sound. If you put on a record from that era, very often you will hear right away what it is. Carcass sounds like Carcass, Bolt Thrower sounds like Bolt Thrower, and so on. Black metal emerged eventually, and it worked the same way, but somewhere along the way, things started going more and more in the same direction. What we have now is just sad. It sounds great, I'm not going to say otherwise, but things are so 'put in place'. Everything is 'perfect' and polished, but the identity is lost. If you put something on today, you have no idea what it is until, maybe, the vocals come in, as long as the vocalist has some distinct voice. I don't know what it is. Maybe people just don't dare to diverge from the norm or something.

"I haven't been immune to this development. I was trailing along in the same footsteps for a while, but I had to step on the brake, eventually. I wish more producers would do the same."

Much of the reason why "everything is fixed and edited to absolute perfection" these days is that it has become "so easy to just go in and fix every single little detail". Stjerna expands: "We have almost endless possibilities to alter what is recorded, but that doesn't mean we should. Back when I started out, you just couldn't. Eventually, that changed drastically, and I, too, started editing the crap out of things, but it sucked the life out of the recordings.

"I still do edits. I still brush up things, but I paint with broader strokes. I spend more time during the tracking to get the result I'm looking for. The ideal is somewhere in between, or even a bit more toward not getting everything absolutely perfect — that can actually be better if you want a really big, atmospheric sound. It depends a little bit on the style, as well."

Stjerna is also wary of overusing sample replacement on drums: "Most of the time, I will blend in a triggered kick with the miked signal. With snare, it's maybe 50-60 percent of the time. It depends a lot on the performance. But I try to pay close attention to maintaining a 'natural sound'. With toms, I'll add in a bit of triggered signal... if someone holds a gun to my head."

Stjerna enjoys providing feedback and pushing groups toward creative growth. Frontman Paul Delaney of New York band Black Anvil confirms that Stjerna is the only man who has managed to preserve the "magic" of his band's demos while giving their records a sense of enormity. "For black and death metal, in my opinion, he is number one. On *Regenesis*, he gave us a new sound, based on how he heard the

new songs. Way more midrange, clarity, and ultimately more life. Having been always used to and comfortable with laying in the cut bass-wise, my tone was crushing, but it may not have been helping the big picture. Adjusting my tone live and finding a happy medium gives everything a breath of fresh air and doesn't compromise any of the punishing factor that I pride myself on." Not only has Stjerna's brilliance as an engineer greatly altered Paul's attack on bass, but Stjerna's output with Funeral Mist, the record *Salvation* in particular, changed Paul's approach to music years before the pair met.

Further Mayhem

All of which brings us back to Mayhem's definitive *De Mysteriis Dom. Sathanas*, the Pytten-produced masterpiece that established Norwegian black metal as a genre. Mayhem are now on the roster of Stjerna's artist management company NBS Production, and Black Anvil accompanied Stjerna on the road for Mayhem's 2017 tour of *De Mysteriis*. Stjerna recorded, mixed, mastered and, along with Mayhem's Teloch, contributed intros for *De Mysteriis Dom. Sathanas Alive* — an album capturing a 2015 performance in Sweden.

When it came to the *De Mysteriis* shows, Stjerna affirms: "It was a matter of replicating what equipment was used and going from there." This included Hellhammer's famed drum sound. "We did similar tuning of the drums. It's very low tuning. At that time, Hellhammer was using a lot of triggers, so we just kind of took away all that and just went over to mics, which is what we're still doing nowadays."

Fortunately, the team was able to find the same kind of amp that Euronymous used: "It's not the very same guitar amp that Øystein used. That one is 'buried by time and dust', I guess. When he died, a lot of his things were just tossed out. It's also made it difficult to figure out what he actually used, and others that he played with back then sadly don't know or can't remember. His [*Marshall*] JCM800 can be seen in several photos, though, so that was less of a mystery. He used various pedals over the years, as well, but we had to experiment a bit more with that. I think we got pretty close in the end."

After *De Mysteriis Dom. Sathanas Alive*, Stjerna and Mayhem recorded the EP *Atavistic Black Disorder / Kommando*, the studio album *Daemon*, and the live record *Daemonic Rites*. Stjerna reveals that "after the *DMDS* tours, we went back to Peavey 6505 for the guitars, which the band have been using for a long time now. But apart from that, I guess we've continued on the same path. With *Daemon*, what I was aiming for was to make it sound like what pops up into your head if someone says 'Mayhem', and I think it's pretty clear what that is in my case — and I think most would agree. I use the same approach live."

At the 2022 edition of the Beyond The Gates festival, Mayhem again reprised *De Mysteriis* with the help of Necromorbus, with a deeply moved Pytten watching.

Reflecting on his career, Stjerna states: "I'm honoured to have worked with a lot of really great bands, and I'm thrilled that people like the stuff that I do. I'm very happy that I have been able to contribute something to the world, and that I have a legacy. That's really the most important thing for me."

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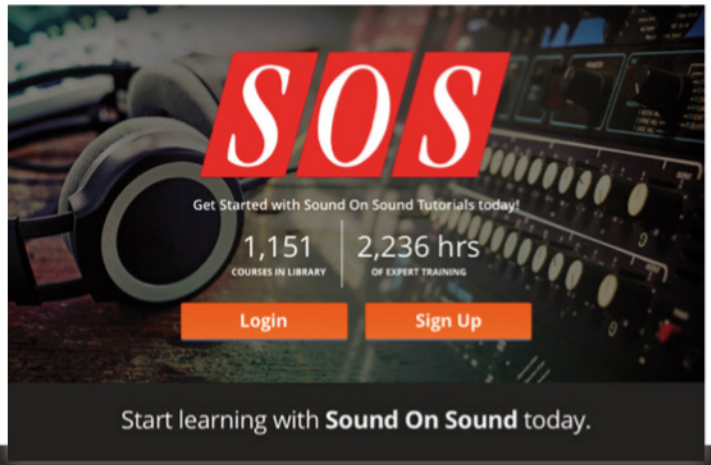

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SAMPLE LIBRARIES

Auddict
Broken Heartstrings Piano
Kontakt Instrument

★★★★½

I've been keeping my eye on Auddict since reviewing Drums Of The Deep in July 2017. Following their impressive debut, this versatile UK company built up a sizeable catalogue of orchestral titles, including Angel Strings (reviewed in *SOS* in March 2021) and the excellent Master Solo Woodwinds, which features a jaw-droppingly realistic legato mode.

For his latest release, Auddict CEO Dorian Marko dusts off his concert pianist chops and invites us to get our hands on his Steinway Model D concert grand. To achieve the requisite soft, tender tone, this magnificent instrument was modified for the sampling sessions: a layer of felt was introduced, the hammers were treated and the soft pedal calibrated by a piano tech. Happily, the re-engineering was reversed after the sampling was completed.

The piano was recorded in Mr Marko's studio from three mic positions using Royer R-121, Coles 4038 and Neumann U87Ai mics. The mid and far positions add a little room ambience, and you can use the built-in reverb to create a more distant concert hall sound. In an interesting departure from standard piano miking, the close mic position was adjusted according to the range being recorded, thus ensuring a super-close, intimate sound across all 88 keys.

After adjusting my keyboard's velocity curve to suit the instrument's dynamic response, I found it responded well to sensitive, improvisatory playing. Though essentially soft and gentle, its overall tone remains clear and fairly transparent, with enough attack to maintain a rhythmic presence. The 16 velocity layers and full length sustains also guarantee a naturalistic response with plenty of dynamic expression.

Front-panel controls include stereo width, percussive hammer and pedal noise, the latter the bane of sound engineers the world over. The Lament setting introduces a per-key capture of the piano's natural resonance which, when used sparingly, gives it more size and body.

Also available are transformative effects such as Echoes (a floaty, subtly modulated long reverb), Haunt (which adds a ghostly upper octave) and the spacey pad-like shimmers Whisper and Hush. There's also a beautiful, ethereal dedicated pad layer and a great Plectrum option which gives you the sound of the piano strings played with a guitar pick, creating a cheerful jangle reminiscent of a Persian santur or pub piano. Having adjusted these settings, you can save your work in one of the 12 preset slots.

I've played a few sampled pianos which broke my heart (not in a good way), but this one brightened my morning. This review may have missed the deadline for Valentine's day, but if your loved one is in the market for a highly playable felted sampled grand, this piano would make the ideal romantic present. Cheaper than a real Steinway but expensive-sounding nonetheless, Broken Heartstrings (9.35GB installed) requires the full version of Kontakt 4 or higher. *Dave Stewart*

$240

www.auddict.com

Naroth Audio
Guitar Odyssey
Kontakt Instrument

★★★★

The first thing to say about Guitar Odyssey is that it isn't about creating conventional guitar sounds, but rather about creating entirely new sounds that use guitar recordings as the initial sound source. Hosted by the latest version of the free Kontakt Player or Kontakt v6.6.1 onwards, the instrument comes with over 7GB of samples and 200 categorised presets. The Odyssey engine allows for the layering of four sounds with individual envelope control or amplitude and filter, though there's also a master envelope control option.

The presets show off an impressive range of soundscapes, pads and aggressive leads as well as some inspiring short melodic sequences, cinematic textures and drones, so if scoring for picture is your thing, there's a lot of potential here. Ambient music composers will also find a lot to love, though composers of more conventional pop music may find some of the offerings a little too esoteric.

The way Guitar Odyssey is set out allows for sounds to be sculpted in a number of different ways, depending on how deeply you want to get into editing. Loading a preset and then changing the samples in the various layers is a good starting point, and from there it is easy to do basic editing such as adjusting envelope and filter settings. The comprehensive range of effects also provides plenty of scope for sound design; these are very easy to manage, with the relevant controls for each effect visible when you click on the relevant FX block. Should you wish to venture deeper, the Movement and Modulation pages offer a huge amount of scope, as does the application of granular processing. This is an instrument that rewards experimentation.

Exploring the presets reveals dreamy pads, ominous drones, melodic patterns and shimmering soundscapes but no twanging guitars. The core samples are the result of heavily processed guitar recordings, the end result rivalling anything that can be generated by pure synthesis and arguably more organic-sounding for that. Many of the sounds incorporate a palpable sense of movement, which gives them a very organic feel, ably demonstrated by preset four, Abandoned Toyshop. There's a huge breadth of cinematic potential here — when it comes to guitar sounds we're definitely not in Kansas anymore. *Paul White*

$199

www.narothaudio.com

Soniccouture
Waterphone
Kontakt Instrument

★★★★★

With its bizarre appearance and endearing-but-creepy sound, the acoustic instrument known as a waterphone has become a firm staple of the horror and soundtrack worlds. Soniccouture have embraced the role of sample capture, while introducing some interesting twists to their reinterpretation in Kontakt form.

Acoustic waterphones are circular instruments with a resonant cavity at the base which is usually filled with a small

amount of water. Extending from this resonator are a number of metal tines, surrounding a handle in the centre. These tines can be struck, bowed or brushed, using fingers, sticks, mallets or a bow. The resulting acoustic output is a vibrant collection of pure harmonics, the pitch and colour of which are altered as the water sloshes around inside the resonator.

This Kontakt instrument library divides into two components, both at around 3GB in stature. The first section is formed from phrases: a cornucopia of the waterphone's greatest hits, exploring shifting harmonics and subtle scrapes, with speed variance from slow and lingering to relatively fast and exaggerated. This adopts a phrase-per-note make-up with editable control over each phrase's playback. You can reverse the sample and alter the start time, while also changing the pitch. Due to its extraordinary sonic purity, it's possible to push the pitch editing to the extreme without losing integrity, which yields some quite amazing effects.

The second instrumental section is labelled Waterphone Unwrapped. As the instrument is recognised as being rather inharmonic, Soniccouture have attempted to provide patching that is playable, at least in a more conventional way than might usually be associated with the acoustic instrument. This really does work to a sizeable degree, though the nature of the sound does mean that you can quickly find your compositional output becoming overwhelmed by the sustained harmonics. Thankfully, there are plenty of control elements that allow the waterphone to be tamed, ranging from an ADSR envelope to 25 different filter colours, which include low-, high-, band-pass and notch filtering, and vowel settings.

The waterphone's method of initial attack can also be dictated with a selection of bows, strikes and the very useful Tuned Accent option. The Harmonic control allows adjustment between the fundamental and the harmonic an octave above, while the on-board LFO can be steered in the direction of the Harmonic control with a separate LFO that controls the rate of the water sloshing!

The Mic page accesses three faders representing the waterphone's recorded signals. These are, specifically, a close-miked signal, contact mic and the resonant cavity containing the water. From this point alone, you can considerably vary the timbral qualities of the instrument.

Professional-grade acoustic waterphones are relatively expensive, which makes the idea of a sample library, created by experts in the field, all the more appealing and cost-effective. It's a beautiful-sounding and impressive Kontakt instrument, in line with the rest of Soniccouture's catalogue of more obscure and inspiring instruments. *Dave Gale*

$119

www.soniccouture.com

Sound Dust
DRIFT003
Kontakt Instrument

★★★★

Requiring the full version of Kontakt, v5.8.1 or above, DRIFT003 uses as its source a 9GB library of voice-like sounds, arranged as 11 multisampled hybrid articulations. From these are woven 180 snapshots and 32 modular stacks.

The majority of the sounds were created from AI vocal generators, though we're told there is a small amount of actual mouth-generated sound in there too. The recordings have been processed and edited to create 11 separate four-octave, multi-velocity articulations. While the resulting sounds can be identified as some type of vocal articulation, reality is most definitely not the focus here. Rather, the aim is to create something new that still suggests human voices, almost as though a vocoder eloped with a multi-effects processor after listening to a Tuvan throat singer serenading a Speak & Spell game.

The results are varied, from chaotic gibbering to choral overtones, drones that mumble away to themselves and humanoid basses. Some make perfectly useful playable sounds, whereas others may work better as drones or musical punctuation. The way the mod wheel is employed in different ways to modify the sound for each articulation adds a further performance dimension.

The core sounds comprise long evolving samples, loops and multiple velocity layers, so you don't hear a lot of repetition. Each of the 11 articulations has its own name and reveals a distinctive sonic character: Free Speech, Host, Dysfunction, Folk Devil, Squeakbox, In Yun, Splutterer, Piano Ghost, Mouth Trumpet, Door and Imaginary Friend. The convolution reverb engine is controlled by the Space knob, which goes from dry to 100 percent wet. The lo-fi antics of the wow and flutter engine can be tempo-sync'ed, as can the tremolo functions, making rhythmic pulsing easy to arrange. There's also a chaos page, where randomisation can be applied individually to Velocity, Tuning, Volume, Pan and Time using five knobs, which can add a further organic dimension.

The DRIFT003 user interface resembles four partly sucked Polo mints stuck to an army blanket, with three tabs at the bottom of the window for selecting the main DRIFT003 view, the Chaos view with its five controls, or the RTFM page, which provides a very brief overview of the controls. The reverb types, shown to the right of the Space knob, are categorised as Large, Medium, Small, Reverse or Spring, with several subcategories for each type, including some useful shimmer treatments. The tremolos and Drift controls can be assigned to MIDI controllers, and of course all the main parameters can be automated, which can result in some wonderfully complex, evolving sounds, especially if both the EQ Morph and Grist controls are moved during the course of a sustained note. Add these to the timbral changes mapped to the mod wheel and there's lots of scope for creating very long, evolving sound beds

Despite the weirdness, I found a lot to like in DRIFT003, with many of the sounds inviting further processing, such as feeding through a granular delay or layering with more conventional pad sounds. As indicated, some of the sounds work well as playable pads, albeit ones that seem to be chattering in a slightly unsettling way, while the Mouth Trumpet articulation in particular makes for some very warm pads and bass sounds. There are also lots of 'ear candy' sounds that would work well layered with other sounds. You won't find any realistic choirs or operatic sopranos here, but if the idea of a talking computer, high on the electronic equivalent of magic mushrooms appeals to your musical sensibilities, then I think you are going to find DRIFT003 a lot of fun. *Paul White*

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Q How do I make live-tracked metal guitars sound sufficiently wide?

I'll soon record a sludge-metal band and I need some advice. They want to record with them all playing in the same room at the same time, but there's only one guitar. How would you deal with that in relation to the final mix's stereo image? I wanted to pan the same guitar L/R, but of course I don't want to make it mono again. Any good techniques?
Anon. via email

Mike Senior, SOS Contributor

Well, there are a few options I'd suggest. The first is to just pan the guitar centrally, then rely on its room ambience to give it some width. However, given the well-established metal trope of wide-panning double-tracked rhythm guitars, I wouldn't expect that to deliver the kind of left-right breadth the band will be looking for. Likewise, you could create some stereo width from the close-miked guitar sound using panned multi-miking, but again I'm not sure that'd create the degree of guitar spread you'd normally expect in metal.

A more promising option should be to mic up two guitar amps with somewhat contrasting/complementary tones, and then hard-pan those in the mix to generate a more obvious stereo spread, and there are a few ways you could implement this. My favourite option would actually be to set up both amps in the room either side of the drum kit, and feed them from the same guitar via a dedicated splitter box. Alternatively, you could just take a DI from the guitar on the way to a single amp, and then simultaneously reamp that in a separate room while tracking — or indeed reamp it after the main tracking session. Now, while it might seem easier to leave the reamping until after tracking, do bear in mind that a more impressive guitar sound on the tracking session will impact on how the band feels, so the extra hassle of implementing the reamping 'live' may be worth it in terms of improving the general vibe on the sessions, and potentially getting better performances as a result. And remember, even in that case, if you take a DI you can always rehash the live-tracked reamped sound later anyway, so there's no need for live reamping to tie your hands too much at mixdown.

Whether you create stereo spread by multi-miking or by reamping, though, you will need to beware of phase-cancellation between the hard-panned mics/layers in mono. Certainly, you should make a point of checking your guitar texture in mono while tracking, so that you avoid any nasty mono-compatibility problems. If you find the sound collapses in mono, try flipping polarity switches on some of the guitar mics in the first instance, or changing the miking distances slightly if polarity inversions aren't helping much.

Radial's BigShot ABY can be used to route one guitar signal to two amps simultaneously (or either one individually) and, importantly, includes an ground-lift function.

Despite all of the above suggestions, though, it's still possible that, by comparison with a lot of commercial metal releases, you might not be able to get a sufficiently impressive stereo spread in this way. If that's the case, my last-ditch solution would be simply to overdub a double-track of the guitar part after the fact, and hard-pan that against the live-recorded guitar. You'll likely need to add a bit of artificial room ambience of some kind to get the overdub to sit comfortably with the live track, but otherwise there shouldn't be any real difficulties in doing that, as long as the player can adequately recreate their part. Honestly, if any metal band with a single guitarist asked me to do a record with them all playing together, this is actually the approach I would plan for as a backstop, simply because I could pretty much guarantee to get an appropriate guitar image that way. But, at the same time, I'd still try to get the best out of multimiking/reamping — if I actually managed to get sufficient width that way, I could then bask in the glory of saving the guitarist from all that manual double-tracking! Always best to keep expectations in check and then over-deliver!

Matt Houghton, Reviews Editor, adds:

While I've nothing to add to the approaches outlined in Mike's reply, which lays out your strategic options very clearly, I thought it might be worth addressing the question of how, on a practical level, you might go about feeding two amps from a single guitar simultaneously. You can use a DI and re-amp together, as he suggests, but if you don't already have those, there are other options, as well as some approaches to avoid. It's important that there's an earth lift to break any ground loops that might cause problems, and that's one reason you can't really 'bodge' this using some sort of Y-cable, and why a stereo effects pedal's left and right outputs might not do the job either. To do it 'properly', you'll want a buffered splitter pedal or a transformer-based ABY pedal; an ABY pedal's footswitches allow you to feed the guitar signal to one amp, the other, or both at the same time, making it a useful utility gadget to have in your kit bag. Various manufacturers make them, and a good example is Radial's BigShot ABY. For stereo or more experimental setups (such as if you want to feed amps, DI boxes and modellers in parallel) the same company offer a device called the Shotgun that can feed four amps simultaneously from one or two input signals, and invert the polarity on any of them.

SOUND ON SOUND

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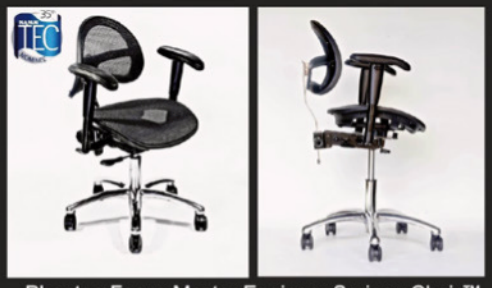
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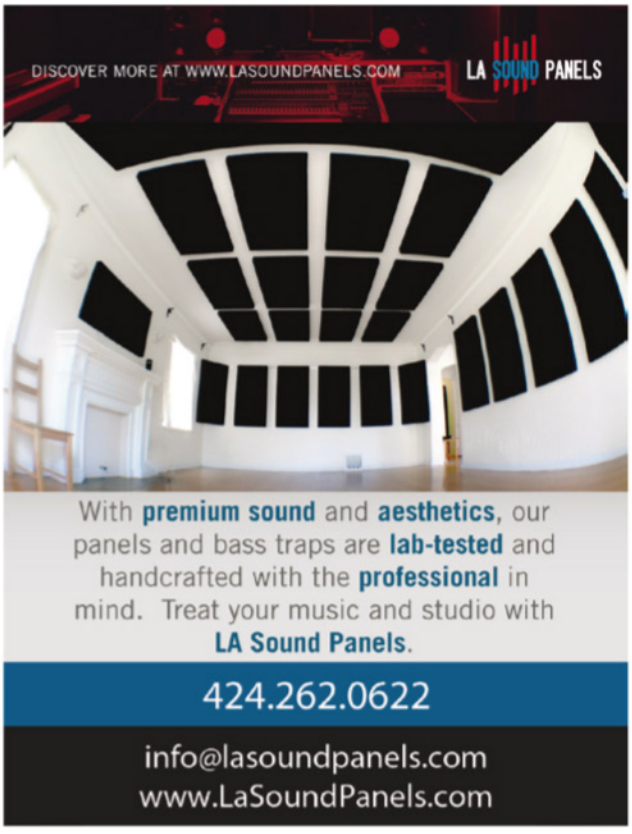

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SOS WHY I LOVE...

MODERN POP PRODUCTION

RINGO JEDLIC

When I first heard Dua Lipa's 2020 record *Future Nostalgia* I was blown away. This was a new smash hit pop record that was good. Really good. And I don't mean this in the backhanded 'I can really respect her as a talented artist and performer' kind of way, or the pretentious 'I think it's an objectively good album' kind of way either (both of which are usually followed by a 'but...'). Being the wannabe hipster and music snob that I am, my immediate reaction should have been to discount Lipa's album as some sort of commercialised product of the culture industry, a cheap postmodern nostalgia trip, or insert whichever depressing insight we often have into the modern pop industry here. The music speaks for itself, however, and *Future Nostalgia* was — at least to my ear — something new and different. Sonically, lyrically, and conceptually this album represented for me the culmination of several trends and new approaches in music-making which make me cautiously hopeful for the future of pop music.

In Robert Strachan's 2017 book *Sonic Technologies*, the author observes a shift that happened in music production throughout the 2010s, whereby an increasing number of top-charting pop songs were being produced on what some people had previously considered non-professional DAWs. Sure, many top artists are still recording in Pro Tools and the software is still used for a lot of tracking and mixing (often in conjunction with other DAWs, as we can see from *Sound On Sound*'s 2023 interview with Rob Bisel about working on SZA's 2022 hit 'Kill Bill') but this speaks to the larger point of Strachan's book: the DAW — which was once just a tool for recording — has become the central creative instrument for music-making. Gone are the days of expensive studios, session musicians, and labels developing artists, and as Strachan suggests, we have entered the era of small, hyper-exclusive teams of hybrid producer-songwriters recording on laptops in hotel rooms and bedroom studios.

This brings me back to Dua Lipa. Reading the *SOS* interview with Ian Kirkpatrick (pictured) about producing for Lipa was an eye-opening look into this new world. Kirkpatrick uses Cubase, works in a home (bedroom) studio, and does everything in the box. He often relies on using plug-ins in strange and creative ways to create the core rhythmic and melodic components of his tracks. Although Lipa's music has dance elements, *Future Nostalgia* isn't electronic. It is a full-blown bubblegum pop hit — a pop hit made/written in the box in Cubase, with vocals recorded on a handheld mic. Kirkpatrick's session breakdown videos on YouTube also demonstrate this in-the-box approach to writing pop; he's constantly fiddling with plug-in parameters and experimenting with different ways of using various effects and software instruments.

Now to get to my ultimate point here, I think that the adoption of the DAW as the central creative instrument in pop music represents a new era of using the studio musically and expressively, In a way, it's reminiscent of the type of experimentation that blossomed in pop recording during the late-'60s multitrack revolution.

An up side to the decline of the pop industry and rise of streaming is that at least producers are given a decent level of creative and sonic freedom, as labels can no longer afford (or are no longer willing to pay for) expensive studios with ageing executives and similarly ageing but worse looking technicians in them breathing "but what's the setup?" down their necks.

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